IRIS-Net

en User Manual

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IRIS-Net

Preface

IRIS-Net - Intelligent Remote & Integrated Supervision - is an efficient PC-program running under Windows. It allows configuring, controlling and monitoring of a complete PA-system from one central location. Control and monitoring up to 250 Remote Power Amps is possible from a single or from several PC's. The IRIS-Net System provides a complete overview of the entire system status at all times plus real-time control of all relevant system parameters. IRIS-Net allows the creation of customized graphical user interfaces to optimally match any application offering easy control and monitoring even for large and complex installations and PA systems.
To get the most out of the IRIS-Net software we recommend that you read through all the topics in this help file carefully.

Notes on The IRIS-Net Documentation

The left hand side of the Online Help allows convenient selection of individual chapters. Chapters that consist of several subchapters carry a book icon, which opens by double clicking on it. Chapters may include reference links to other chapters that provide related information. Double clicking such a link opens the corresponding chapter.

Basics of IRIS-NET

IRIS-Net System Requirements

Processor	Dual core CPU
OS	Windows 7 (32-Bit or 64-Bit), Windows 8 or Windows 8.1 Project Generator or Dx46/DSP 600: The .Net framework (3.5sp1 or higher) is required. Dante: Windows 7 64-Bit recommended
RAM	2 GB (more is recommended)
Hard Disk	2 GB of free space
Video	1024x768, High Color (16 Bit)
Network	Ethernet port and/or 1 USB port per 100 amplifiers Audio network: Gigabit Ethernet port

Installation

This manual guides the user through the installation process of IRIS-Net using the following operation systems: Windows 9x, NT, 2000, XP and 7.

1. In Windows click on Start > Run. The Run dialog box appears.
2. Click on the Browse... button. The Browse... dialog box appears.
3. Select the directory in which the IRIS-Net installation file is located.
4. Select "setup.exe" and click on the Open button. The IRIS-Net installation program starts. Follow the instructions of the installation program.
5. When the installation is finished you can start IRIS-Net from Windows via Start > All Programs > IRIS-Net > IRISNet.

IRIS-Net Directory Structure

This section provides information about IRIS-Net directories and the contained files which, after installing the software package, are stored on your computer's hard disk. IRIS-Net's default installation path is C:\Program Files\IRIS-Net\§ Version §. All subfolders are created in this directory. During installation, defining any other installation path is possible as well. Subfolders and files will then be created at the new location.

Files installed in the main folder
Amongst others, the main IRIS-Net program folder holds the following files:

Filename	Description
IRISnet.exe	This is the executable program file. A double click on the file's icon starts the IRIS-Net application.
IRIS_readme.pdf	ReadMe document offering additional information.

Subdirectories

\Bitmaps.

This folder contains several picture files in bitmap format, e.g. loudspeakers, racks, front views of different devices, logos, etc. which, for example, can be used to design the graphical representation of a PA-system within an IRIS-Net project. Creating your own bitmaps and saving them in this folder is possible as well.

\Controls.

This directory holds all IRIS-Net controls. Controls are composed of system files that are of no direct relevance for the user and bitmap graphics. The latter define the looks of a control. Creating and adding custom bitmaps which then represent your own specific controls is possible.

\Documentation.

This folder contains manuals and documents related to IRIS-Net plus subdirectories that hold the manuals for several devices and systems.

\Driver.

This folder holds Windows drivers for different interfaces and applications. At the moment, it also includes two subdirectories. Prior to using/installing drivers and interfaces, please make sure to consult the corresponding help files.

\Firmware.

This directory contains the firmware files of all the devices that can be used with the IRIS-Net software.

\Help.

This folder holds all help files. These files have the extension .htm.

\Projects.

This directory is for saving your IRIS-Net project files. We recommend creating a separate project folder for every new project under the path \Projects and to save all the files related to an individual project in the corresponding project folder. Strictly following this convention greatly facilitates transferring a project from one PC to another because you only need to copy the project folder including its entire contents and save it under the same name on the other PC under the following path \IRIS-Net\Projects.
The IRIS-Net package includes example projects ”Demo System Small” and ”XLC Demo System” which can be used to form the basis for creating your own projects. The two projects are also good examples to get familiar with IRIS-Net’s hierarchical file structure and the way a project should properly be saved.

\RCM-24 Presets.

The Default Presets F01 and U01 - U08 are located in this folder. Default presets are for initializing a power amplifier as soon as it is included in an IRIS-Net worksheet via drag-and-drop. The subfolders contain factory-presets for speaker cabinets.

\RCM-26 Presets.

The Default Presets F01, F02, O01, O02 and U01 - U06 are located in this folder. Default presets are for initializing a power amplifier as soon as it is included in an IRIS-Net worksheet via drag-and-drop. The subfolders contain factory-presets for speaker cabinets.

\Speaker Files.

The speaker files for different loudspeaker cabinets are located in this folder. Speaker files contain measured transfer functions (frequency response and phase response) of loudspeaker systems. IRIS-Net lets you import and display this data. This allows the user to see the frequency and phase response of set filter, x-over, level, phase and delay parameters as well as the resulting acoustical transfer function of DSP parameters in use with the measured speaker data.

\Speaker Settings.

This folder holds the configuration files for several different loudspeaker cabinets. Each model has its own directory, which houses the files for each loudspeaker cabinet of the particular model. Contained in these files are factory predefined and optimized settings - equalization (PEQ), x-over, level trim, alignment delay, and compressor/limiter - for the corresponding loudspeaker. IRIS-Net allows importing and applying those speaker settings to a single amplifier channel or to complete groups of amplifiers. So, basically at the push of a button, the user can establish optimal settings for the connected speaker cabinet.

\Tools.

This folder holds the IRIS-Net Project Generator. For using the IRIS-Net Project Generator execute the file setup.exe.

\User Controls.

The preconfigured control panels contained in this directory can be utilized in IRIS-Net projects. The control panels consist of one or more controls and a bitmap picture to provide graphical display of the panel. The functions of each control come factory-preconfigured, so that activating a control panel is a matter of simply connecting it to the desired devices or groups. IRIS-Net also provides the possibility for the user to create custom user controls, which should be save in this folder as well.
All other subdirectories contain IRIS-Net system files, which are of no further significance for the user. As a matter of fact, the opposite is the case. Altering the contents of these subfolders is categorically not advisable.

Introduction

Creating A New Project

An IRIS-Net project includes the complete configuration as well as the parameter, operation and monitoring settings for a Remote Power Amplifier System. Creating and saving various user control panels is possible. Individual password protection per page is provided.
Configuration of A Remote Power Amplifier System Using a PC-CAN-Interface

1. Start The IRIS-Net Software. IRIS-Net starts creating a new project and opens an empty "Configuration Page" worksheet.
2. Saving. Name your new project before saving (File Save). Creating a separate folder for each project in the directory \IRIS-Net\Projects\ is strongly recommended. All files associated with a specific project (e.g. bitmaps) need to be located within that particular folder. Periodically saving the project is generally good advice.
3. Creating Devices. Clicking the right mouse button within the Worksheet opens a Configuration Dialog. Click on Add Devices... using the left mouse button, drag the required power amps out of the device list and drop them in the worksheet area. Specify the number of devices to be created as well as the start address. The devices appear together with a PC-icon on the computer screen when hitting OK. The symbol of the PC represents the control-PC and the CAN-interface.
4. Network Connection. If your computer is already linked to a remote power amp network, you are able to go ONLINE. The network dialog window indicates which power amps are connected/linked and whether the connection is okay or not. Now is the moment to decide, whether you want to continue with the configuration in ON-LINE or OFF-LINE mode.
5. Creating Groups. Open the Configuration Dialog by clicking the right mouse button while the cursor is located over the worksheet window. Use the left mouse button and click on Add Control...and drag a Group element into the worksheet. Right clicking on the Group icon followed by a click on Properties opens the group properties dialog. Now, you are able to make the desired settings. It is also here, that you can specify which panel opens by double-clicking the group icon. Creating groups is always recommended, when several power amps or power amp channels are to be controlled or configured simultaneously. Typical examples are:

– System Group: Switching On / Off, Change Presets, Master EQ
– HF / LF Groups: Uniform configuration for X-Over and speaker system equalization
– Left / Right Groups: Common operation features of the PA system (left / right)

6. Loading Preset Data into Groups. While being in the Group Properties dialog you can specify that double-clicking the group icon automatically opens the Setup & Control window. The IMPORT PRESET and EXPORT PRESET soft keys are located on the DSP FLOW DIAGRAM page under the DSP tab. Clicking IMPORT PRESET opens a selection box that allows choosing from the Preset files. Selecting and opening a file transmits its stored parameters to all power amps associated with that group. Afterward, you need to save these setting in a free User Memory (U02...U08).
7. Setting Group Parameters. Opening the Setup & Control window by double-clicking a group icon lets you use the DSP Dialogs to set all parameters of all power amps within one group at the same time. This ensures that all parameter values of all devices within a group are identical.
8. Creating Additional Pages. Open the Configuration Dialog by clicking the right mouse button while the cursor is located over the worksheet window. Use the left mouse button and click on Add Layer. This creates a new page (layer). Now, you have to name it (e.g. Control Page). IRIS-Net allows the use of up to 32 layers.
9. Designing User Control Panels. The IRIS-Net software allows the creation of various control panels, which can be customized to suit the needs of a project and the needs of the users. Freely programmable control elements, bitmaps, text boxes and scripts are provided for you to do so. Selecting Add Control in the Configuration Dialog (right click on the worksheet) opens a list of control elements. Drag the needed elements into the worksheet area

and place them as desired. To select and place bitmaps and text boxes, follow the same procedure accordingly (Add Bitmap..., Add Textbox...). In case you need several similar control groups, start with creating one group. Once you are finished with that particular group, select the entire group by dragging a highlight box around it and Copy/Paste it. Alternatively you can save the complete group as a User Control (Save User Control menu command) and reuse the User Control in various other projects.

10. Programming User Control Panels. Control elements of control panels can be used to control or to indicate various power amp parameters. A click with the right mouse button onto the control element opens the corresponding configuration dialog. Choose Administrate Connections and select the desired connection for the control element (power amps, groups, other control elements) from the connections list. Select Modify Properties for setting/ modifying the control element's properties. Click in the line on the right next to function and select one or several parameters to be controlled from the parameter listing. If needed, set additional control element properties accordingly. Afterwards, make sure to test the control element's functions.
11. Programming Power Amplifier Parameters. A double click on a power amplifier icon opens the Amplifier Control Panel. The Amplifier Control Panel allows naming power amps and channels, switching power amps on / off, setting level controls and mutes, and assigning and monitoring input or output signals to the monitor bus. Click on the SET button to open the Setup & Control Window. This window includes five different pages: Config & Info, DSP, Speaker, Load, Supervision & Test. "Config & Info" provides information about power amplifier type, firmware revision and configuration. In addition you are able to program the power-on delay, enter power amp and channel names, and specify functions of the GPIO port. Furthermore, programming job functions (control functions, which are sent and received via the CAN-network) is also possible. "DSP" allows setting all DSP parameters and saving them in presets. Recalling presets, import (read from file) and export (save to file) is possible as well. The DSP parameters include Master EQ, Master Delay, Routing, Channel EQ, X-Over with Level Trim and Alignment Delay as well as Dynamic functions including compressors and limiters. On the "Speaker" page you are able to select loudspeaker files with measured frequency and phase responses and assign them to individual power amplifier channels. This response plots can be displayed in EQ and X-Over windows to allow optimized parameter settings. "Load" provides information about measured output voltages and currents and the resulting calculated load. You can set the limits for the allowable impedance range and start impedance tests of a defined frequency range. Impedance graphs can be saved as reference measurements with tolerance spread. "Supervision & Test" provides indication of different fault types. You are able to select which failures result in a fault messages. Activating the pilot tone function is also performed on this page. Also provided is a test tone generator for manual testing.
12. Password Protection. Each page (layer) can be protected by an individual password. Programming different access rights is possible as well. Open the password dialog in the Menu Configuration, Passwords.... Enter a password for the system administrator (probably you) and add new users (New User) if needed. You have to enter the user names, passwords and access rights. Specifying at least an administrator password and protecting the configuration page against unwanted access is generally recommended.

Configuration Of A Remote Power Amplifier System Using a NetMax N8000 System Controller

1. Start The IRIS-Net Software. IRIS-Net starts creating a new project and opens an empty "Configuration Page" worksheet.
2. Saving. Name your new project before saving (File Save). Creating a separate folder for each project in the directory \IRIS-Net\Projects\ is strongly recommended. All files associated with a specific project (e.g. bitmaps) need to be located within that particular folder. Periodically saving the project is generally good advice.
3. Creating N8000 System Controller. Clicking the right mouse button within the Worksheet opens a Configuration Dialog. Click on Add Devices... using the left mouse button, drag a N8000 out of the device list and drop them in the worksheet area. Specify the number of devices to be created. The devices appear together with a PC-icon on the computer screen when hitting OK. The symbol of the PC represents the control-PC and the NCP-interface.

4. Creating Power Amps. Clicking the right mouse button within the Worksheet opens a Configuration Dialog. Click on Add Devices... using the left mouse button, drag the required power amps out of the device list and drop them in the worksheet area. Specify the number of devices to be created as well as the start address. The interface of the N8000 is preselected. The amplifiers appear on the computer screen when hitting OK.
5. Network Connection. If your computer is already connected to the N8000, you are able to go ON-LINE. Check the N8000 manual for details about Ethernet configuration. The network dialog window indicates which power amps are connected/linked and whether the connection is okay or not. Now is the moment to decide, whether you want to continue with the configuration in ON-LINE or OFF-LINE mode.
6. Creating Groups. Open the Configuration Dialog by clicking the right mouse button while the cursor is located over the worksheet window. Use the left mouse button and click on Add Control...and drag a Group element into the worksheet. Right clicking on the Group icon followed by a click on Properties opens the group properties dialog. Now, you are able to make the desired settings. It is also here, that you can specify which panel opens by double-clicking the group icon. Creating groups is always recommended, when several power amps or power amp channels are to be controlled or configured simultaneously. Typical examples are:

- System Group: Switching On / Off, Change Presets, Master EQ
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8. Setting Group Parameters. Opening the Setup & Control window by double-clicking a group icon lets you use the DSP Dialogs to set all parameters of all power amps within one group at the same time. This ensures that all parameter values of all devices within a group are identical.
9. Creating Additional Pages. Open the Configuration Dialog by clicking the right mouse button while the cursor is located over the worksheet window. Use the left mouse button and click on Add Layer. This creates a new page (layer). Now, you have to name it (e.g. Control Page). IRIS-Net allows the use of up to 32 layers.
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13. Password Protection. Each page (layer) can be protected by an individual password. Programming different access rights is possible as well. Open the password dialog in the Menu Configuration, Passwords.... Enter a password for the system administrator (probably you) and add new users (New User) if needed. You have to enter the user names, passwords and access rights. Specifying at least an administrator password and protecting the configuration page against unwanted access is generally recommended.

How to edit an existing project?

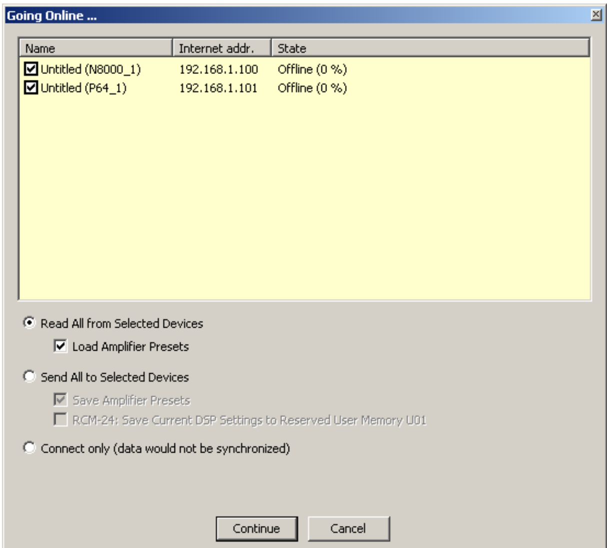
This paragraph explains how to edit a project that already exists. Due to the fact that IRIS-Net provides the possibility for different users to have different access permissions assigned within a single project, editing options might be highly restricted by the Administrator.

1. Start the IRIS-Net program. IRIS-Net starts, creates a new project and opens a new worksheet named "Layer 1".
2. Open an already existing project. Select File > Open. The Open dialog box appears. Select the desired project directory and then the according project file (extension .ds) and click on the Open button.
3. Enter the password (optional). The Login dialog box appears automatically if the project that you are about to open is password protected. Enter the Administrator user's password in the Enter Password: text field.
4. Edit the project. Now you are able to edit the project, i.e. change devices, controls and displays/indicators of the current project. Besides changing parameters and configurations, you can also add new or remove elements. Save the project (optional). If you have altered the project and want to save the changes, select File > Save to save the project under the same name. In case you want to save the project under a new name, select File > Save As... and enter a new file name.

Going On-Line

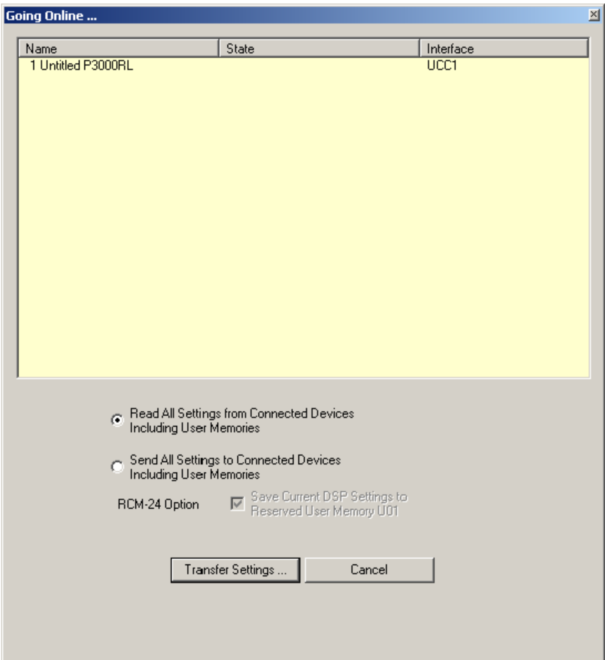
IRIS-Net allows working off-line or on-line. The two modes differ from each other in the way of selecting executable operations. Upon opening, the IRIS-Net application is automatically in offline mode. This mode allows the creation and editing of a project. The dialog that appears when pressing the "On-line" button in the "Button Bar" differs, depending on the interfaces used in the project. The Ethernet On-line Dialog appears fro each Ethernet interface used in the project. The CAN On-line Dialog is displayed for each CAN interface that is used.

Ethernet On-line Dialog



Element	**Description**
Name	Name of the device. The checkbox in front of the name of a device allows selecting the unit as a connecting device. A connection will be established only with selected devices.
Internet addr.	The IP address of the device in IRIS-Net. Connection fails, if the IP address specified in IRIS-Net differs from the actual IP address of the unit.
State	Indicates the current connection status of the device.
Read All from Selected Devices	Reads all settings of connected device and transfers them to IRIS-Net.
Load Amplifier Presets	Additionally reads the presets of all connected Remote Amplifiers.
Send All to Selected Devices	Writes the DSP configuration and all settings into memory of connected devices.
Save Amplifier Presets	Additionally writes the presets of all Remote Amplifiers.
RCM-24: Save Current DSP Settings to Reserved User Memory U01	Additionally writes the current DSP settings into the User Memory U01 of all power amplifiers with RCM-24 Remote Control Module installed.
Connect only (data would not be synchronized)	No synchronization; i.e. data (DSP configuration, settings, presets) is not transferred to the devices nor read from the devices.

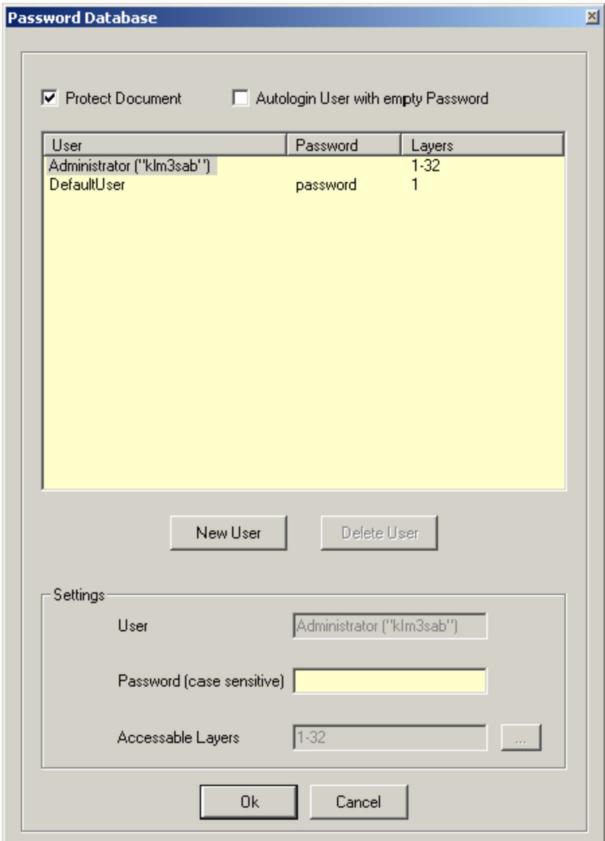
CAN On-line Dialog



Element	Description
Name	Name of the Remote Amplifier.
State	Indicates the current connection status of the device.
Interface	Name of the PC-to-Remote Amplifier interface.
Read All Settings from Connected Devices Including User Memories	Reads all settings and presets of the Remote Amplifier(s).
Send All Settings to Connected Devices Including User Memories	Writes all settings and presets into memory of the Remote Amplifier(s).
Save Current DSP Settings to User Memory U01	Additionally writes the current DSP settings into the User Memory U01 of all Remote Amplifiers with RCM-24 Remote Control Module installed.

Project Password Protection

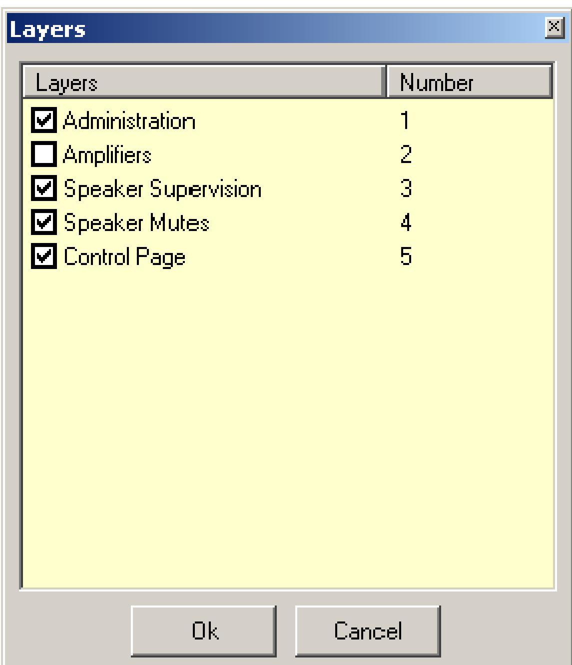
The "Password Database" Window allows creating new user accounts. It is possible to grant or to deny a user access to specific layers within a project.



Element	Description
Protect Document	A project's password protection can be activated or deactivated.
Autologin User with empty Password	If a user has a "blank" password, on opening, the project is automatically loaded with his access rights.
User	Username.
Password	The user's password. This column is only visible when the administrator account is being used.
Layers	The layers that can be accessed by the user.
New User	Creates a new user account and adds it to the List of Users.
Delete User	Deletes the user(s) that have been selected/marked in the List of Users.
User	This field allows entering the user's name. **CAUTION: A user name may not contain a","(comma).**
Password	This field allows entering a password for the user. **CAUTION: A user name may not contain a","(comma).**

Accessable Layers	This field allows entering the numbers that correspond to the layers which are accessible for the user. Several layers have to be separated by commas. Using hyphens for separation indicates that the layers are interrelated.
...	Opens the "Layers" Window which allows selecting those layers that are accessible for a user.
Ok	Keeps the changes that have been made and closes the window.
Cancel	Discards the changes that have been made and closes the window.

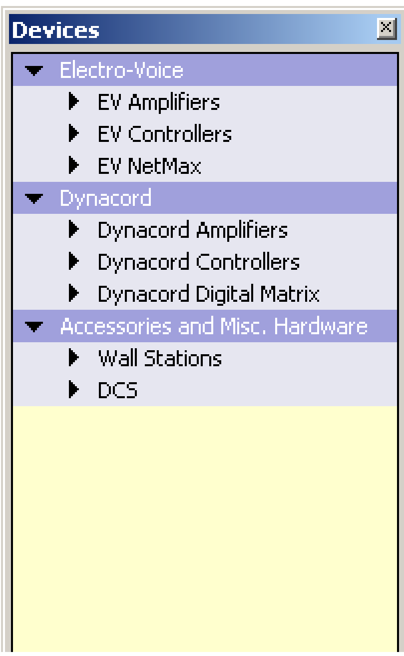
The Layers dialog allows selecting the layers that are accessible for the selected user.



IRIS-Net Objects

Object List

The Object List provides access to all necessary elements for creating and editing IRIS-Net projects. By default IRIS-Net opens with the Object Bar being located on the left-hand side of the worksheet. However, placing the Object Bar at any position within the IRIS-Net worksheet is possible as well. Displaying or hiding the Object Bar is possible via View menu. The following illustration shows the Object Bar with the Controls listing being open. Additional categories are: Devices, Interfaces, User Controls, Bitmaps and Textboxes.



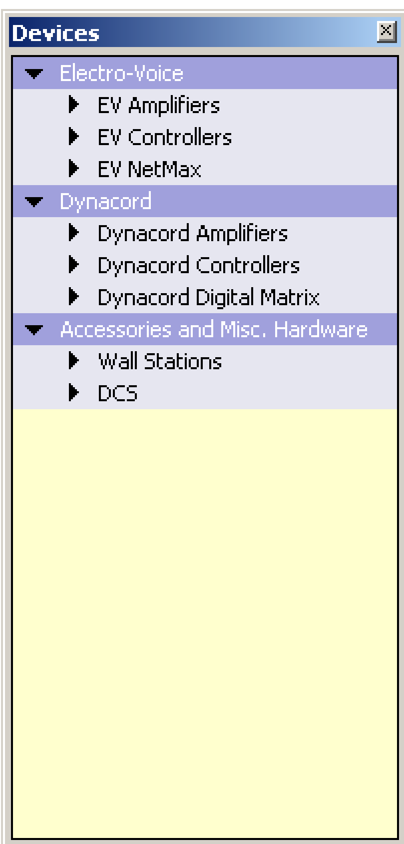
Different Object Bar Categories

Category	Description
Devices	The Devices category lists IRIS-Net devices, which can be included in the worksheet via drag and drop.
Interfaces	The Interfaces category lists all ports and interfaces which IRIS-Net supports. You can drag and drop Interfaces into the worksheet and afterwards establish connections with existing Devices. Some Interfaces are automatically created and assigned.
Controls	The Controls category lists all available controls.
User Controls	The User Controls category lists pre-defined control panels that allow easy creation of project and client-specific user and display panels. Creating custom User Controls and adding them to the list is possible as well.
Bitmaps	This category lists custom-designed and pre-defined Bitmaps, which the user can select and include in an IRIS-Net project.
Textboxes	Select one of the Textboxes listed in this category to place a text message in the IRIS-Net worksheet. The dialog box for entering the text opens automatically.

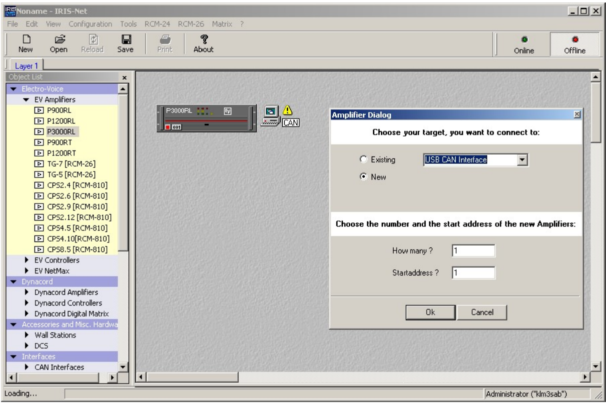
Object List categories can also be accessed from the Configuration menu or the Configuration Dialog Box (click right mouse button in the worksheet) and be displayed in an individual window. For detailed explanation of elements and their utilization, please refer to the according chapters.

Adding Devices

Devices are units and components, which can be configured, controlled and monitored within IRIS-Net. Examples are: Remote Amps, signal processors, speaker controllers, corresponding modules and peripheral devices. The list of devices that IRIS-Net supports expands constantly.
All currently available IRIS-Net Devices can be accessed from the Objects Bar (listed under the category Devices) or from the separate Devices window, which opens after clicking on the item Add Device. Add Device is available from the IRIS-Net Configuration menu or from the contextual menu within the IRIS-Net worksheet.
The following illustration shows the Objects Bar with the Devices listing being open. The devices are categorized in groups to provide a better overview. Opening and closing the group lists is possible by clicking on the arrow icon at the right. The contents of the devices list depends on the actually installed IRIS-Net version and on the devices that are actually available, i.e. it can differ from the lists shown.



To add devices to an IRIS-Net project, first select the desired device in the Object Bar (or from the Devices window) and then drag and drop it into the worksheet. A dialog box opens, which lets you specify device-related settings such as amount of desired devices, address range, and interfaces. The illustration shows how to add a P3000RL power amp to an IRIS-Net project. The following table lists the possible settings that are available in the Amplifier Dialog.

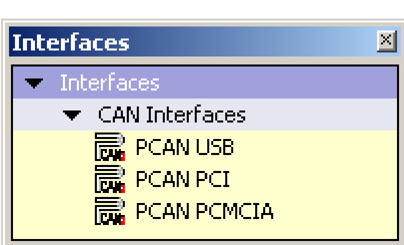


Element	Description
Existing	If an appropriate interface (between PC and device) already exists in the IRIS-Net project, you can select that interface from the drop-down field.
New	Select an interface from the drop-down field, to automatically include it in the IRIS-Net project.
How many ?	Specify the amount of units of the selected type.
Startaddress ?	The CAN-Bus address of the selected device. If more than a single device is to be added to the project, addresses are assigned in ascending order.

The dialog box closes upon hitting the OK button and the previously specified amount of devices appears in the IRIS-Net worksheet. Devices can be selected, dragged around and repositioned at will. Double clicking on a device opens the corresponding configuration dialog box. For further detail, please refer to the according chapters.

Adding Interfaces

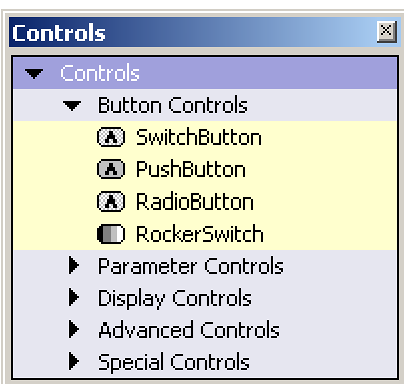
Interfaces serve as gateways between the PC that runs the IRIS-Net application and devices that can be configured, controlled and monitored from within IRIS-Net, e.g. interfaces between PCs and a CAN-Bus.
Interfaces that are supported by IRIS-Net are listed under the Interfaces category in the Objects Bar and in the separate Interfaces window. This window opens when you choose Add Interface from the IRIS-Net Configuration menu or from the contextual menu within the IRIS-Net worksheet. To include an interface in an IRIS-Net project you just have to select the desired interface in the Interfaces list of the Objects List (or in the Interfaces window) and drag & drop it over into the worksheet. Newly added interfaces are by default not connected to any device. To assign devices to an interface, select Administrate Connections form the contextual menu of a device and select the desired interface.
The following illustration shows the Objects List with the Interfaces list being open. The listed interfaces are divided into groups to provide a better overview. Opening and closing the group lists is possible through clicking on the arrow icon in the top right corner. The content of the interfaces list depends on the actually installed IRIS-Net version, i.e. it may differ from the list as shown in the following illustration.



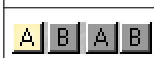
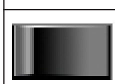
Element	Description
PCAN USB	Interface (UCC1) between USB port of a PC and CAN bus.
PCAN PCI	PCI board serves as interface between PC and CAN bus.
PCAN PCMCIA	PCMCIA card serves as interface between PC and CAN bus.

Adding Controls

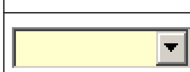
IRIS-Net Controls are located in the Objects List under the Controls category and in the separate Controls window. The window opens when you choose Add Control from the IRIS-Net Configuration menu from the contextual menu within the IRIS-Net worksheet. Several different controls like buttons, switches, displays, Window Controls, Panels, groups, etc. are available. The listed controls are divided into groups to provide a better overview. Opening and closing the group lists is possible through clicking on the arrow icon in the top right corner. Using drag & drop you can include controls in the IRIS-Net worksheet to design a suitable user and or indicator panel for your project. An example of how to configure a control is provided below.
The following illustration shows the Objects List with the Controls list being open. The content of the controls list depends on the actually installed IRIS-Net version and on the devices that are currently available, i.e. it may differ from the list as shown in the following illustration. The table also provides a short description for each control.

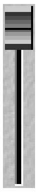
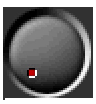
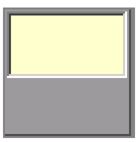
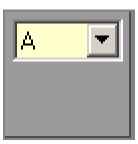
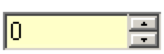


Button Controls

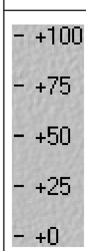
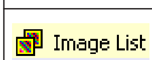
Element	Name	Description
	Push Button	Pushbutton for executing a function or for opening an IRIS-Net dialog box
	Radio Button	Several pushbuttons that trigger each other, e.g. for switching or changing functions or parameters
	Switch Button	Switch button for switching between two states, e.g. On/Off
	Rocker Switch	Rocker switch for switching between two states, e.g. On/Off

Parameter Controls

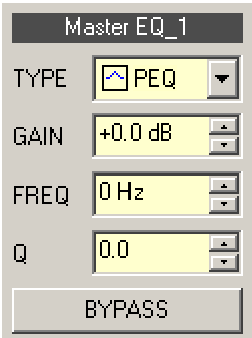
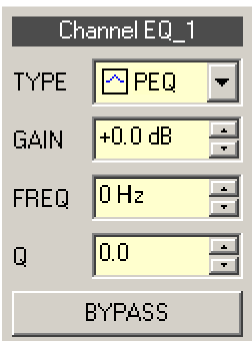
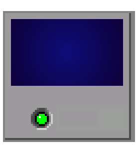
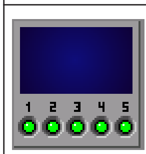
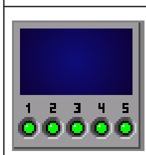
Element	Name	Description
	Combo Box	Windows Combo Box for selecting parameters or functions from a list

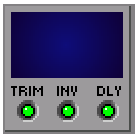
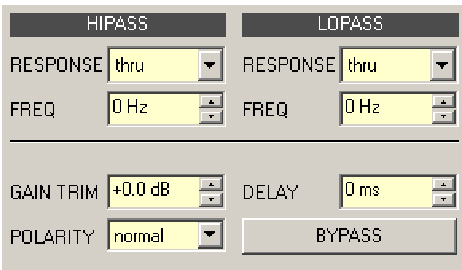
	Edit	Edit field for entering and displaying values or text, e.g. names
	Fader	Fader for setting parameter values
	Bar Graph	Bar graph display horizontal for graphical indication of parameter values, e.g. operational states
	Knob	Knob for setting parameter values
	Mini Edit	Label block for displaying names in signal flow diagrams
	Mini Routing	Routing block for setting the output routing in signal flow diagrams
	Spin Edit	Windows Spin Edit for entering values via Up/Down selection buttons or direct editing

Display Controls

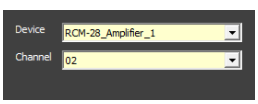
Element	Name	Description
	Display Value	Display field for displaying values or text, e.g. names
	LED	LED indicator for signaling operational states or errors
	Meter	Bar graph display vertical for graphical indication of parameter values, e.g. meter instruments
	Scale	Scale for labeling vertical bar graph indicators
	Image List	Image List for switching between different images

Advanced Controls

Element	Name	Description
	MasterEQ	PEQ user panel for a single or a group of Master EQs
	ChannelEQ	PEQ user panel for a single or a group of Channel EQs
	Mini Delay	Delay block for displaying delay settings
	Mini Dynamics	Dynamic block for displaying settings of dynamic processors
	Mini MasterEQ	Master EQ Block for displaying Equalizer settings
	Mini ChannelEQ	Channel EQ Block for displaying Equalizer settings

	Mini XOver	X-Over Block for displaying Frequency Crossover settings
	XOver	X-Over user panel for a single or a group of Frequency Crossover blocks

Special Controls

Element	Name	Description
	Group	Group for grouping several similar components, e.g. devices, channels, etc.
	Dante Control	Dante Control for a DM-1 Dante Modul (e.g. of a N8000) and RCM-28 remote amplifiers. Possible connections: TG-5.ChA, TG-5.ChB, TG-7.ChA, TG-7.ChB, H2500.ChA, H2500.ChB, H5000.ChA, H5000.ChB, N8000.DSP.DanteIn_x.Chy, P64.DSP.DanteIn_x.Chy

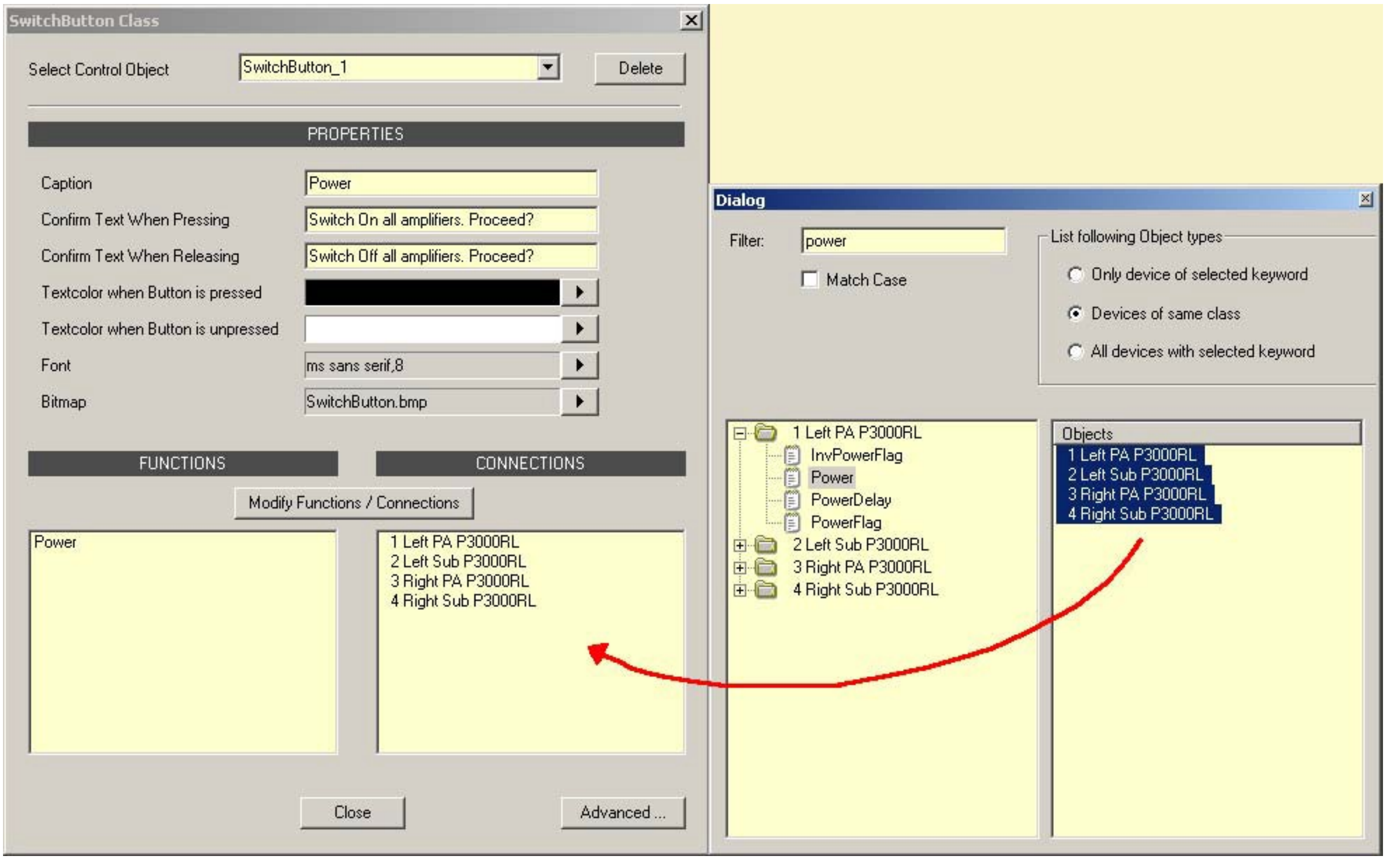
How to configure a Control

For using a Control it must be configured with one or more parameters. A parameter always consists of two components, a "Function" and a "Connection". If one of the two components is missing, the Control will not work. In this example a Switch button shall be used to simultaneously switch several amps of a project on or off.

1. Drag and drop a Switch button from the list into the IRIS-Net worksheet.
2. Click with the right mouse button on the Switch button and select Modify Properties from the switch button’s contextual menu. The Switch Button Class window appears.
3. Edit the button’s appearance and labeling in the Switch Button Class window.
4. Click on the Modify Functions / Connections button. The Modify Functions & Connections window appears.
5. Enter power in the Filter text field (using a filter is optional). Only objects that include the parameter power are being displayed in the functions list, which is shown in the left part of the window. The functions are grouped by devices used in the project. Within the devices the functions a grouped for comfortable browsing. Grouping within a device can be deactivated via the entry "Function Structure" in the groups context menu.
6. Click at the symbol "+" in front of one amplifier in order to see the list of functions.
7. Select Power from the functions of the amplifier (see picture below).
8. All amps that are included in the current project are being displayed in the Objects list, which is shown in the right part of the window.

9. Select all desired amps in the Objects list and drag them over into the Connections field of the Switch Button.

Class window



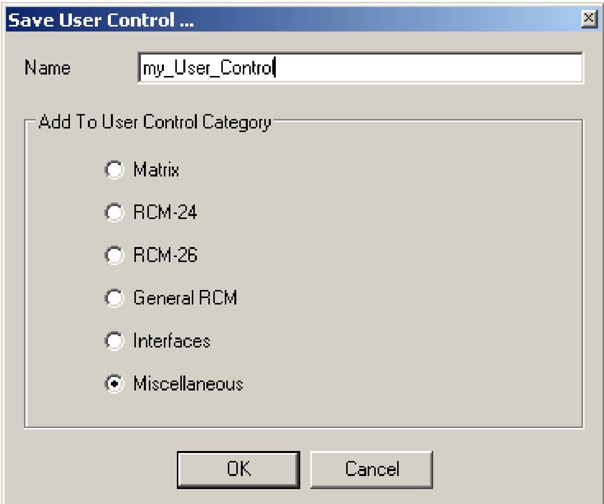
10. Close the Modify Functions & Connections and Switch Button Class windows.

This completes the configuration of the switch button. It can now be used to switch the selected amps on or off.

Adding User Controls

User Controls are links between a single or several Control Elements and a single or several Bitmaps. In addition to using the User Controls that are included in the IRIS-Net package you can also create custom User Controls and save them to disc. IRIS-Net’s preconfigured User Controls are to be found in the Objects Bar category User Controls and in the separate User Controls window. This window opens when you choose Add User Control from the IRIS-Net Configuration menu or from the contextual menu within the IRIS-Net worksheet. Using Drag & Drop you can select User Controls from the list and include them in the IRIS-Net worksheet to create an adequate user and display panel. An example of how to configure a User Control is provided below.

The User Controls listing is divided into several categories. When creating a new User Control the category of the User Control can be selected in the Save User Control dialog.

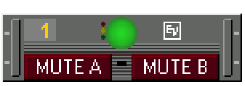
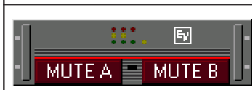
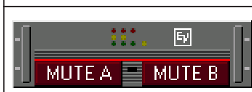
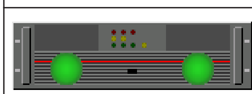
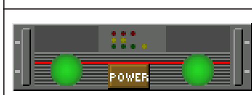


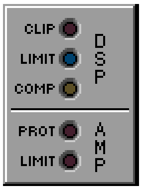
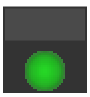
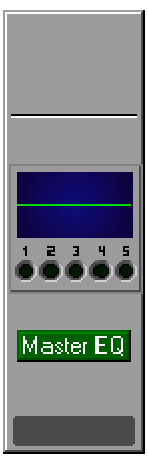
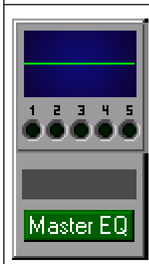
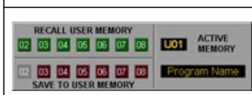
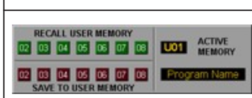
The Locally Used category holds all User Controls, which are used in the current project. The Globally Available category holds all User Controls that are stored in the IRIS-Net User Controls folder. The content of the list depends on the actually installed IRIS-Net version and on the devices that are currently available, i.e. it may differ from the list as shown in the following illustration. The table provides a short description for each User Control.

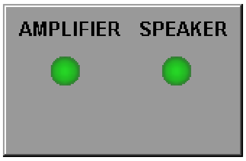
RCM-24 User Controls

Following sets of connections are used in following table:

– RCM-24-Amp: P900RL,P1200RL,P3000RL,P900RT,P1200RT
– RCM-24-Amp-Channel: P900RL.ChA, P900RL.ChB, P1200RL.ChA, P1200RL.ChB, P3000RL.ChA, P3000RL.ChB, P900RT.ChA, P900RT.ChB, P1200RT.ChA, P1200RT.ChB

Picture	Name	Description
	Amp_MUTE+Supervision	The MUTE A and MUTE B buttons activate ChA.Mute or ChB.Mute. In the text field you can set the amp's CAN-address. In Standby or Protection mode the LED lights red. Otherwise, it lights green. Possible Connections: RCM-24-Amp.
	Amp_MUTE_Panel	The MUTE A and MUTE B buttons activate ChA.Mute or ChB.Mute. A click on the amp itself opens the Amplifier Control Panel. Possible Connections: RCM-24-Amp.
	Amp_MUTE_Panel_02	The MUTE A and MUTE B buttons activate ChA.Mute or ChB.Mute.
	Amp_Supervision_01	In Standby or Protection mode the LEDs light red. Otherwise, they light green. LEDs can be labeled. Possible Connections: RCM-24-Amp.
	Amp_Supervision_02	In Standby or Protection mode the LEDs light red. Otherwise, they light green. LEDs can be labeled. Use the POWER button to switch the amp on or off. Possible Connections: RCM-24-Amp.

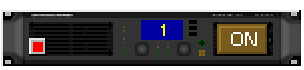
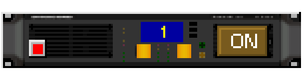
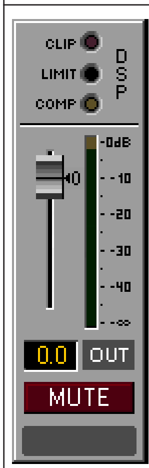
	Group_LEDs_01	The three LEDs on the top indicate clipping (CLIP) or whether the limiter (LIMIT) or the compressor (COMP) of the DSP is active. The two LEDs on the bottom indicate that the unit entered Protection (PROT) mode or that the amp's limiter (LIMIT) has been activated. Possible Connections: RCM-24-Amp.
	Labelled_LED_01	The LED lights red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line) or when the unit is in Standby or Protection mode. Otherwise, the LED is black. Possible Connections: RCM-24-Amp.
	Labelled_LED_03	The LED lights red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line) or when the unit is in Standby, Protection or MUTE mode. Otherwise, the LED lights green. Possible Connections: RCM-24-Amp.
	Master_EQ_Panel_01	Equalizer panel with label field for the RCM-26 Remote Amp. Use the Master EQ button to open the Master EQ dialog. Possible Connections: RCM-24-Amp-Channel.
	Master_EQ_Panel_02	Equalizer panel with label field for the RCM-26 Remote Amp. Use the Master EQ button to open the Master EQ dialog. Possible Connections: RCM-24-Amp-Channel.
	Memory_Panel_01	Load user memory presets 2...8 or save to user memory presets 3...8. Overwriting user memory preset 2 is not possible. Number and name of the currently active user memory is being indicated. Possible Connections: RCM-24-Amp.
	Memory_Panel_02	Load user memory presets 2...8 or save to user memory 2...8. Number and name of the currently active user memory is being indicated. Possible Connections: RCM-24-Amp.

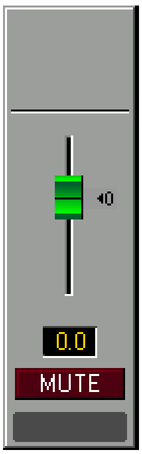
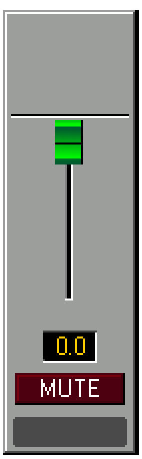
	ShelvingEQ_Freq_Gain_02	Equalizer panel with label field for the RCM-24 Remote Amp. Pre-set for band 5 of the channel equalizer. Equalizer type, frequency, gain and BYPASS can be set. Possible Connections: RCM- 24-Amp.
	System_Supervision_01	The amplifier LED lights red when in Standby or Protection mode. Otherwise, the LED lights green. The speaker LED lights red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line). Otherwise, the LED lights green. Possible Connections: RCM-24-Amp.

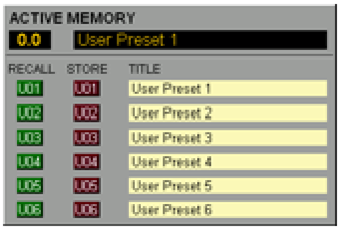
RCM-26 User Controls

Following sets of connections are used in following table:

– RCM-26-Amp: TG-5, TG-7, H2500, H5000
– RCM-26-Amp-Channel: TG-5.ChA, TG-5.ChB, TG-7.ChA, TG-7.ChB, H2500.ChA, H2500.ChB, H5000.ChA, H5000.ChB

Picture	Name	Description
	PowerH_Panel_01	In Online mode the LED light green. Otherwise, the LED light red. The LEDs below the CAN address field light green / yellow when the amplifiers is Power / Standby mode. ON button for switching the amp's power on or off. Possible Connections: RCM-26-Amp.
	PowerH_Panel_02	In Online mode the LED light green. Otherwise, the LED light red. The LEDs below the CAN address field light green / yellow when the amplifiers is Power / Standby mode. ON button for switching the amp's power on or off. Additionally two LEDs for VU data of amplifier inputs. Possible Connections: RCM-26-Amp.
	RCM-26_Group_Panel_01	These three LEDs indicate clipping (CLIP) or whether the limiter (LIMIT) or the compressor (COMP) of the DSP is active. Fader and MUTE buttons for cont- rolling and LED bar graph meter for monitoring the amp’s output channel. Clicking onto the ”0” marking resets the fader to 0 dB. Possible Connections: RCM-26-Amp-Channel.

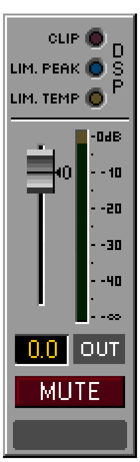
	RCM-26_Master_EQ_Panel_01	Equalizer panel with label field for the RCM-26 Remote Amp. Use the Master EQ button to open the Master EQ dialog. Possible Connections: RCM-26-Amp- Channel.
	RCM-26_Master_Panel_01	Fader and MUTE button for controlling the amplifier. Clicking onto the "0" mar- king resets the fader to 0 dB.
	RCM-26_Master_Panel_02	Fader and MUTE button for controlling the amplifier.
	RCM-26_Memory_Display_01	Number and name of the currently active factory/user/ owner memory is being indicated. Possible Connections: RCM-26-Amp.
	RCM-26_Memory_Panel_01	Load user memory presets 1...6 or save to user memory 1...6. Number and name of the currently active user memory is being indicated. Possible Connections: RCM-26-Amp.
	RCM-26_Memory_Panel_02	Load user memory presets 1...6 or save to user memory 1...6. User memory 1 is protected and can not be overwritten. Number and name of the currently active user memory is being indicated. Possible Connections: RCM-26-Amp.

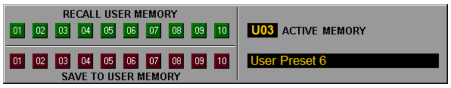
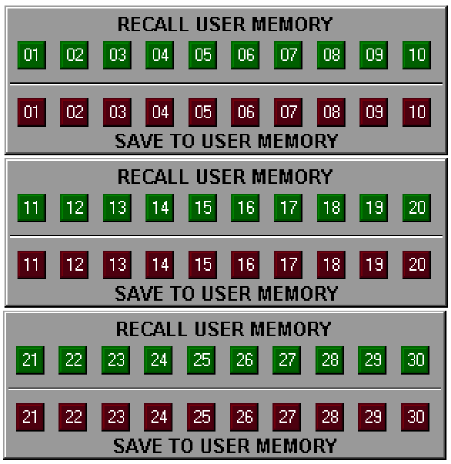
	RCM-26_Memory_Panel_04	Load user memory presets 1...6 or save to user memory 1...6. Number and name of the currently active user memory is being indicated. Editing preset title is possible. Possible Connections: RCM-26-Amp.
	RCM-26_Supervision_Flags_01	CAN address and name of amplifier is indicated. The Temp LED lights red when temperature of the the amplifier is to high. Otherwise, the LED lights green. The Prot A/B LED lights red when amp channel A or channel B is in Protection mode. Otherwise, the LED lights green. The A/B LoZ LED lights red at the occurrence of the load at the amplifier's output is to low (a shorted line). Otherwise, the LED lights green. The A/B hiZ LED lights red at the occurrence of the load at the amplifier's output is to high (a open line). Otherwise, the LED lights green. The white arrow buttons opens the Supervision & Test or Load dialog of the amplifier. Possible Connections: RCM-26-Amp.

RCM-28 User Controls

Following sets of connections are used in following table:

– RCM-28-Amp: TG-5, TG-7, H2500, H5000
– RCM-28-Amp-Channel: TG-5.ChA, TG-5.ChB, TG-7.ChA, TG-7.ChB, H2500.ChA, H2500.ChB, H5000.ChA, H5000.ChB

Picture	Name	Description
	RCM-28_Group_Panel_01	These three LEDs indicate clipping (CLIP) or whether the peak limiter (LIM. PEAK) or the TEMP limiter (LIM. TEMP) of the DSP is active. Fader and MUTE buttons for controlling and LED bar graph meter for monitoring the amp’s output channel. Clicking onto the ”0” marking resets the fader to 0 dB. Possible Connections: RCM-28-Amp-Channel.
	RCM-28_Memory_Display_01	Number and name of the currently active factory/user/owner memory is being indicated. Possible Connections: RCM-28-Amp.

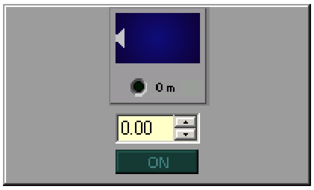
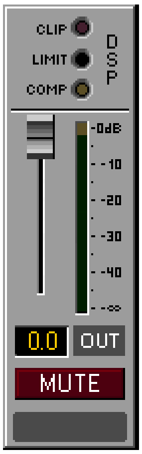
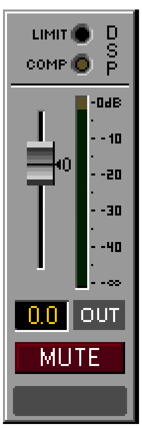
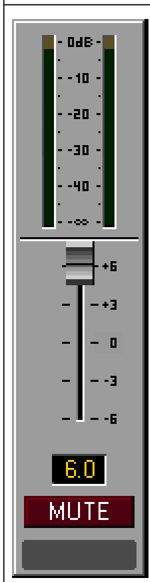
	RCM-28_Memory_Panel_01	Load user memory presets 1...10 or save to user memory 1...10. Number and name of the currently active user memory is being indicated. Possible Connections: RCM-28-Amp.
	RCM-28_Memory_Panel_02, 03 and 04	Load user memory presets 1...10 (or 11..20 or 21..30) or save to user memory 1...10 (or 11..20 or 21..30). Possible Connections: RCM-28-Amp.

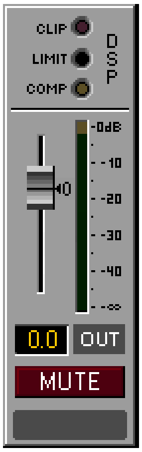
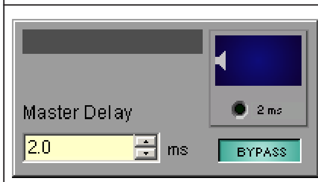
RCM General User Controls

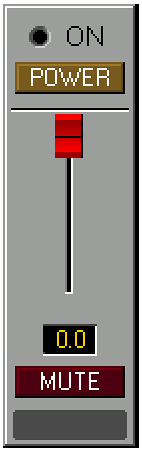
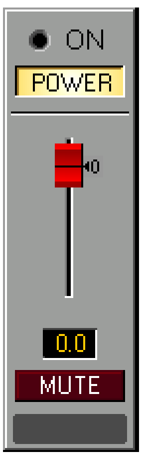
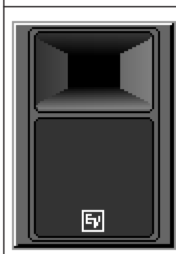
Following sets of connections are used in following table:

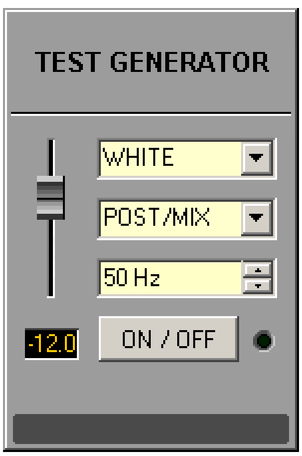
- RCM-24-Amp: P900RL,P1200RL,P3000RL,P900RT,P1200RT
- RCM-24-Amp-Channel: P900RL.ChA, P900RL.ChB, P1200RL.ChA, P1200RL.ChB, P3000RL.ChA, P3000RL.ChB, P900RT.ChA, P900RT.ChB, P1200RT.ChA, P1200RT.ChB
- RCM-26-Amp: TG-5, TG-7, H2500, H5000
- RCM-26-Amp-Channel: TG-5.ChA, TG-5.ChB, TG-7.ChA, TG-7.ChB, H2500.ChA, H2500.ChB, H5000.ChA, H5000.ChB

Picture	Name	Description
	Delay_Panel_01	Delay value and bypass for the channel delay (X-Over) of a remote amp. Possible Connections: RCM-24-Amp-Channel, RCM-26-Amp-Channel
	Delay_Panel_02	Delay value and bypass for the master delay of a remote amp. Possible Connections: RCM- 24-Amp-Channel, RCM-26-Amp-Channel

	Delay_Panel_03	Delay value and bypass for the master delay of a remote amp. Possible Connections: RCM- 24-Amp-Channel, RCM-26-Amp-Channel
	Group_Panel_01	These three LEDs indicate clipping (CLIP) or whether the limiter (LIMIT) or the compressor (COMP) of the DSP is active. Fader and MUTE buttons for controlling and LED bar graph meter for monitoring the amp’s output levels (maximum of both channels). Possible Connections: RCM-24-Amp, RCM-26-Amp
	Group_Panel_02	These two LEDs indicate whether the limiter (LIMIT) or the compressor (COMP) of the DSP has been activated. Fader and MUTE button for controlling the amplifier. The LED bar graph meter is for monitoring the amp’s output levels (maximum of both channels). Clicking onto the ”0” marking resets the fader to 0 dB. Possible Connections: RCM-24-Amp, RCM-26- Amp
	Group_Panel_03	Indicates the output levels of the two channels. Fader and MUTE button for controlling the amplifier. Possible Connections: RCM-24-Amp, RCM-26-Amp

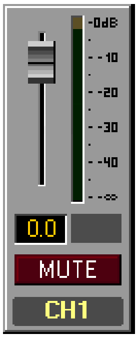
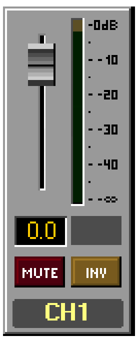
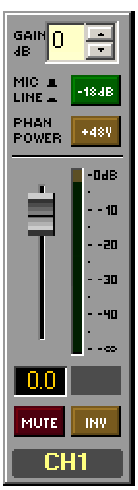
	Group_Panel_GainTrim_01	These three LEDs indicate clipping (CLIP) or whether the limiter (LIMIT) or the compressor (COMP) of the DSP is active. Fader and MUTE button for controlling the amplifier. The LED bar graph meter is for monitoring the amp's output levels (maximum of both channels). Clicking onto the "0" marking resets the fader to 0 dB. Possible Connections: RCM-24- Amp, RCM-26-Amp
	Group_POWER_01	Power-on/off a group of amps with safety dialog when powering off. Possible Connections: RCM-24-Amp, RCM-26-Amp
	Labelled_MUTE_02	MUTE button with label field. Possible Connections: RCM-24-Amp-Channel, RCM-26-Amp- Channel, N8000.DSP.AnalogIn.ChX, N8000.DSP.AnalogMicIn.ChX, N8000.DSP.Analog Out. ChX, N8000.DSP.AutoMixer.ChInX, N8000.DSP.AutoMixer.ChOutX, N8000.DSP.CobraNetIn.ChX, N8000.DSP.CobraNetOut.ChX, N8000.DSP.DigitalIn.ChX, N8000.DSP.LSpkBlock.ChX, N8000.DSP.Matrix.InputX, N8000.DSP.Matrix.OutputX, N8000.DSP.MatrixRouter.InputX, N8000.DSP.MatrixRouter.OutputX, N8000.DSP.Mixer.ChInX, N8000.DSP.Mixer.ChOutX, N8000.DSP.PriorityMatrix.InputX, N8000.DSP.PriorityMatrix.OutputX, N8000.DSP.XOver.ChX
	Labelled_POWER_01	POWER button with label field. Possible Connections: RCM-24-Amp, RCM-26-Amp
	Master_Delay_02	Delay control for the RCM-24 Remote Amp with label field. The delay interval can be entered, BYPASS button and indication (graphical and numerical) of the set delay. Possible Connections: RCM-24-Amp-Channel, RCM-26-Amp-Channel

	Master_Panel_01	Switches the amp’s power on or off with safety dialog when powering off. Fader and MUTE button for controlling the amplifier. Possible Connections: RCM-24-Amp, RCM-26-Amp
	Master_Panel_02	Switches the amp’s power on or off with safety dialog when powering off. Fader and MUTE button for controlling the amplifier. Clicking onto the „0“ marking resets the fader to 0 dB. Possible Connections: RCM-24-Amp, RCM-26-Amp
	Rack_Panel_3Hu_Label_ MUTE	Rack panel with MUTE button and label field. Possible Connections: RCM-24-Amp, RCM- 26-Amp
	Sb121_State_01	The LED in the center of the loudspeaker symbol lights red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line) or when the unit is in Protection or MUTE mode. Otherwise, the LED is invisible. Possible Connections: RCM-24-Amp-Channel, RCM-26- Amp-Channel
	Sx300_State_01	The LEDs in the loudspeaker symbol light red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line) or when the unit is in Protection or MUTE mode. Otherwise, the LED is invisible. Possible Connections: RCM-24-Amp-Channel, RCM-26-Amp-Channel

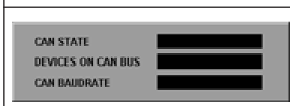
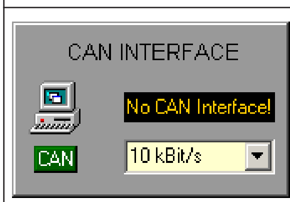
	System_POWER_MUTE_01	Power-on/off and mute a single or a group of amps. Possible Connections: RCM-24-Amp, RCM-26-Amp
	System_Supervision_02	The amplifier LED lights red when at least one of the errors selected in the dialog Supervision & Test occurs. Otherwise, the LED lights green. The speaker LED lights red at the occurrence of the load at the amplifier's output going outside the range set by the minimum and maximum impedance values (a open or shorted line). Otherwise, the LED lights green. Possible Connections: RCM-24-Amp, RCM-26-Amp
	Testgenerator_01	Test tone generator panel with label field for the Remote Amplifier. Possible Connections: RCM-24-Amp, RCM-26-Amp

Matrix User Controls

Picture	Name	Description
	Dante_Panel_01	Selection of Dante device and Dante channel, MUTE button and editable channel description. Indication of LINK status, Signal and 0 dB. Possible Connections: N8000.DSP.DanteIn.ChX

	Level_Panel_01	Fader and MUTE button for controlling and LED bar graph meter for monitoring a input/ output level. Possible Connections: N8000.DSP.AnalogIn.ChX, N8000.DSP.AnalogMicIn.ChX, N8000.DSP.AnalogOut.ChX, N8000.DSP.AutoMixer.ChInX, N8000.DSP.AutoMixer.ChOutX, N8000.DSP.DigitalIn.ChX
	Level_Panel_02	Fader, MUTE button and INV button for controlling and LED bar graph meter for monitoring a input/output level. Possible Connections: N8000.DSP.AnalogIn.ChX, N8000.DSP.AnalogMicIn.ChX, N8000.DSP.AnalogOut.ChX, N8000.DSP.AutoMixer.ChInX, N8000.DSP.AutoMixer.ChOutX, N8000.DSP.DigitalIn.ChX
	Level_Panel_03	Fader, MUTE button and INV button for controlling and LED bar graph meter for monitoring a microphone input level. Additionally Gain, MIC/LINE button and Phantom Power button. Possible Connection: N8000.DSP.AnalogMicIn.ChX

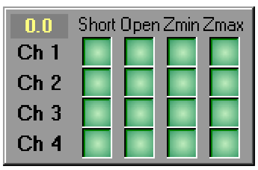
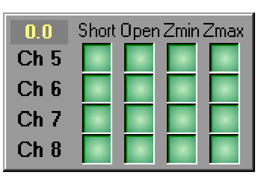
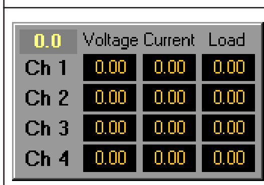
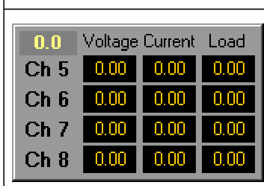
Interface User Controls

Picture	Name	Description
	CAN_Interface_State_01	Shows the momentary state of the CAN bus, the amount of devices connected to the bus and the cur- rent transfer rate. Possible Connections: UCC1
	CAN_Interface_State_02	Shows the momentary state of the CAN bus and the current transfer rate. Use the CAN button to open the CAN Interface dialog. Possible Connections: UCC1

Miscellaneous User Controls

Picture	Name	Description
	Labelled_SPEAKER_01	General loudspeaker with label. Possible Connections: - none -

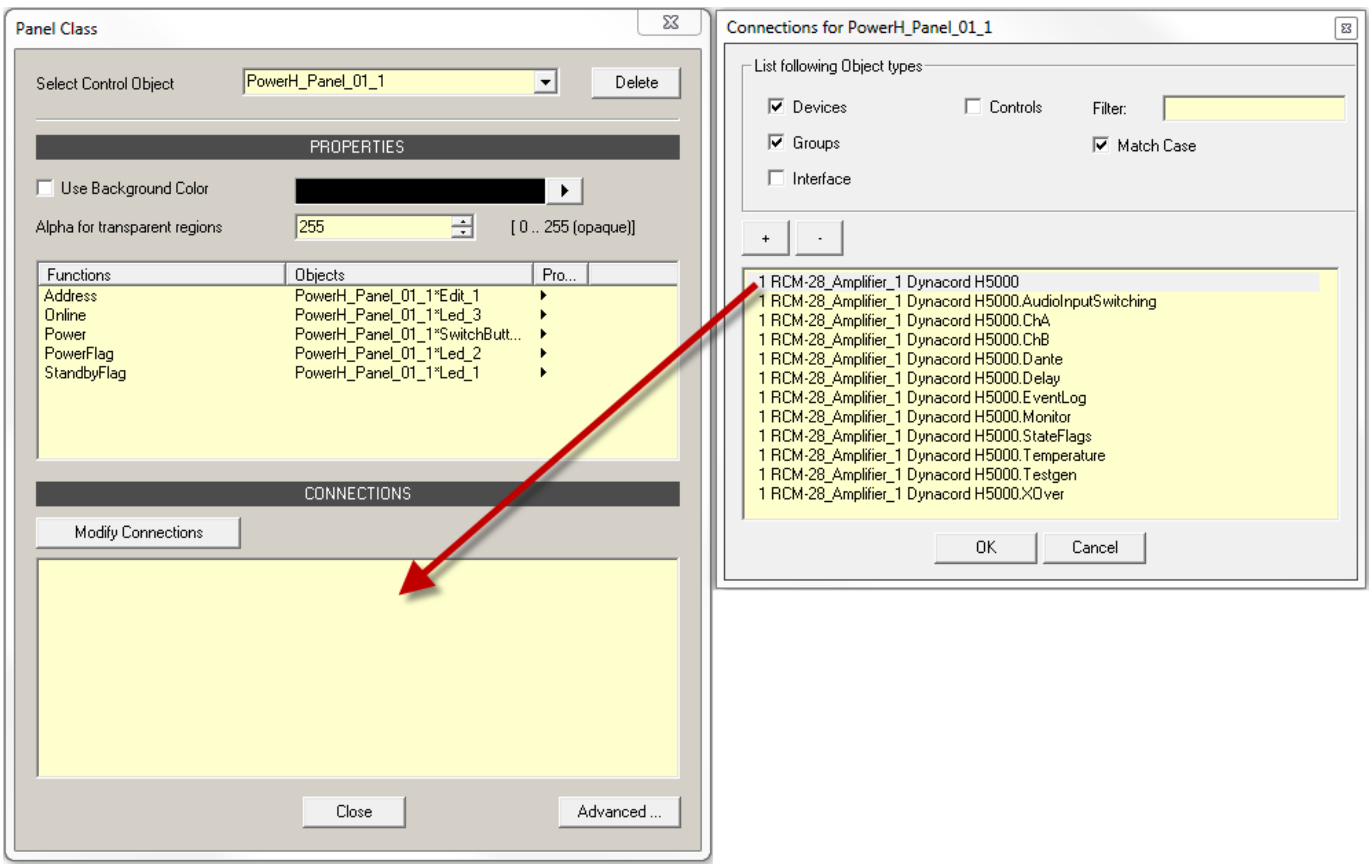
EOL User Controls

Picture	Name	Description
	EOL_LED_Panel_01	The Short LED lights red if the voltage at the amplifier‘s output 1, 2, 3 or 4 is below the EOL VOLTAGE threshold set in the Supervision & Test tab of the amplifier. Otherwise, the LED lights green. The Open LED lights red if the current at the amplifier‘s output 1, 2, 3 or 4 is below the EOL CURRENT threshold set in the Supervision & Test tab of the amplifier. Otherwise, the LED lights green. The Zmin LED lights red at the occurrence of the load at the amplifier's output 1, 2, 3 or 4 is to low (below the LOW THRESH set in the Supervision & Test tab of the amplifier). Otherwise, the LED lights green. The Zmax LED lights red at the occurrence of the load at the amplifier's output 1, 2, 3 or 4 is to high (above the HIGH THRESH set in the Supervision & Test tab of the amplifier). Otherwise, the LED lights green. Possible Connections: CPS4.5, CPS4.10, CPS8.5, DSA 8405, DSA 8410, DSA 8805
	EOL_LED_Panel_02	The Short LED lights red if the voltage at the amplifier‘s output 5, 6, 7 or 8 is below the EOL VOLTAGE threshold set in the Supervision & Test tab of the amplifier. Otherwise, the LED lights green. The Open LED lights red if the current at the amplifier‘s output 5, 6, 7 or 8 is below the EOL CURRENT threshold set in the Supervision & Test tab of the amplifier. Otherwise, the LED lights green. The Zmin LED lights red at the occurrence of the load at the amplifier's output 5, 6, 7 or 8 is to low (below the LOW THRESH set in the Supervision & Test tab of the amplifier). Otherwise, the LED lights green. The Zmax LED lights red at the occurrence of the load at the amplifier's output 5, 6, 7 or 8 is to high (above the HIGH THRESH set in the Supervision & Test tab of the amplifier). Otherwise, the LED lights green. Possible Connections: CPS8.5, 8805
	EOL_U_I_Load_Panel_01	Indicates the current voltage, current and load at the amplifier‘s output 1, 2, 3 or 4. Possible Connections: CPS4.5, CPS4.10, CPS8.5, DSA 8405, DSA 8410, DSA 8805
	EOL_U_I_Load_Panel_02	Indicates the current voltage, current and load at the amplifier‘s output 5, 6, 7 or 8. Possible Connections: CPS8.5, DSA 8805

How to configure a User Control

The example shows how an PowerH_Panel_01 is used to control an amplifier.

1. Use Drag & Drop to include an PowerH_Panel_01_1 in the IRIS-Net worksheet.
2. Click with the right mouse button on the Panel select Administrate Connection from the contextual menu. The “Connections for PowerH_Panel_01_1” window appears.
3. Check the Devices checkbox, in this case you could uncheck all other checkboxes. You can use the Filter in the Connections dialog to filter the list of available objects.
4. Select the desired amplifier from the list and drag it over into the Connections field within the Panel Class window.

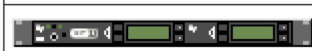
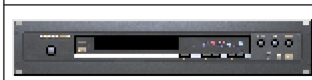
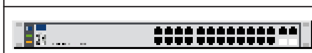


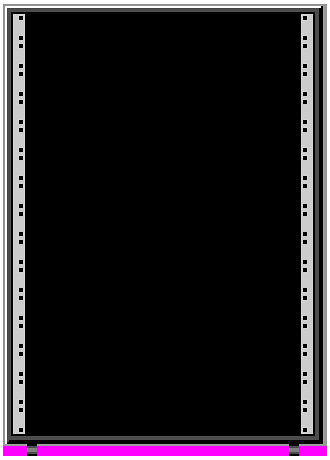
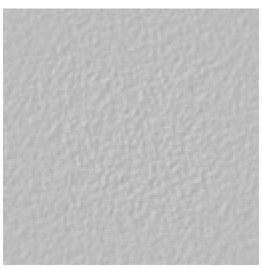
5. Close the “Connections for PowerH_Panel_01_1” window.
6. You can edit the appearance of the User Control (e.g. setting a background color or transparency) in the Panel Class windows.

This completes the configuration of the Amp_MUTE_Panel. It is now ready for use.

Adding Graphics

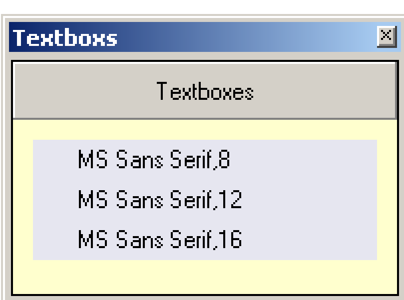
IRIS-Net graphics are available from the Bitmaps category in the Objects List and from the separate Bitmaps window. The window opens when selecting the Add Bitmap command in the IRIS-Net Configuration menu or by using the contextual menu in the IRIS-Net worksheet. The list is divided into two categories. The category Locally Used lists all Bitmaps that are being utilized in the project that is currently open. The category Globally Available lists all Bitmaps, which are stored in the IRIS-Net Bitmaps Folder.
Using custom designed bitmap graphics (in file format bmp, jpg or png) within IRIS-Net projects along with the ones that come with the software package is possible. The color RGB=255,0,255 (HTML=ff00ff) is reserved for defining areas within bitmap graphics that appear transparent within IRIS-Net (e.g. Rack_15HU). You can copy new Bitmaps into IRISNet’s Bitmaps Folder or you can add a new graphic to the IRIS-Net worksheet using Drag & Drop. Just select the desired file in the Windows Explorer and drag it over into the worksheet.

Picture	Name
	CDR1000
	CD_Player
	HP Procurve Switch 2626
	Panel_grey_166x105

	Rack_15HU
	Rack_Panel_1HU
	view_greybkgnd

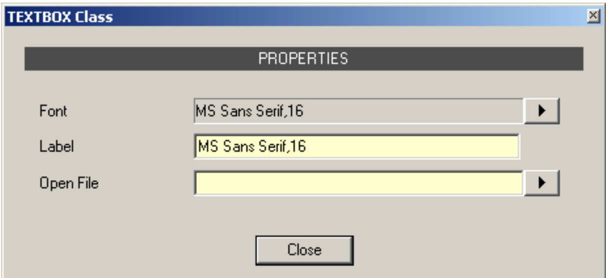
Adding Textboxes

Textboxes, which can be used to create text labels in an IRIS-Net project, are to be found under the Objects Bar category Textboxes and in the separate Textboxes window. This window opens when you choose Add Textboxes from the IRISNet Configuration menu or from the contextual menu within the IRIS-Net worksheet. The window lists a selection of textboxes with different pre-set font sizes. To include a textbox in a project you just have to drag the box out of the list and drop it into the IRIS-Net worksheet.
The following illustration shows the Textboxes window. The table also provides a short description for each item listed.



Element	Description
MS Sans Serif, 8	Textbox using Sans Serif font size 8.
MS Sans Serif, 12	Textbox using Sans Serif font size 12.
MS Sans Serif, 16	Textbox using Sans Serif font size 16.

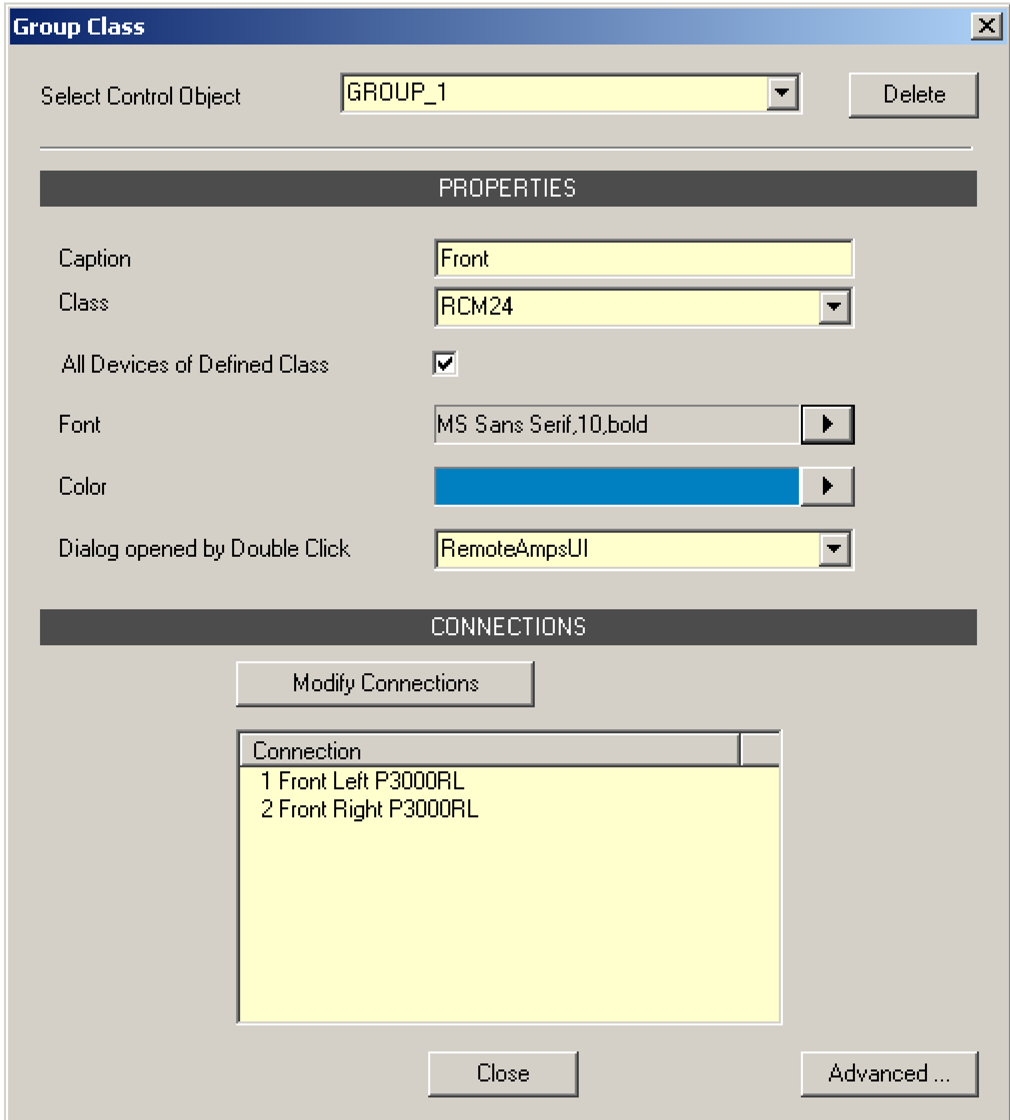
The entry Modify Properties in the contextual menu of a textbox opens the TEXTBOX Class dialog box.



Element	Description
Font	Opens the Edit Textbox window, which allows editing the text contents and appearance of a textbox.
Label	Indicated text.
Open File	Opens a file selection box to create a link to an external file from the textbox. Double clicking on the textbox within the IRIS-Net-worksheet executes the previously selected file, i.e. the operating system starts the program that is associated with the file and then opens the selected file.

Using Groups

IRIS-Net provides Groups as special control elements. Groups are used for combining a variety of objects. Actions applied to a group are applied to all objects of the group. An object can be included in various groups at the same time. Function-specific groups provide a very convenient way to carry out actions that always apply to a specific group of objects.

Symbol	Dialog
	

Element	Description
Select Control Object GROUP_1	Selects the group control to be edited in the dialog.
Delete	Deletes the currently selected group.
Caption	The label displayed on the group icon in the IRIS-Net worksheet.
Class	Selects an object class. The drop down list displays all classes with at least one object contained in the current IRIS-Net works- heet.
All Devices of Defined Class	All objects of the object-class currently selected in the dropdown field “class” are added to the group.
Font	Sets the font used for the group icon label in the IRIS-Net worksheet.

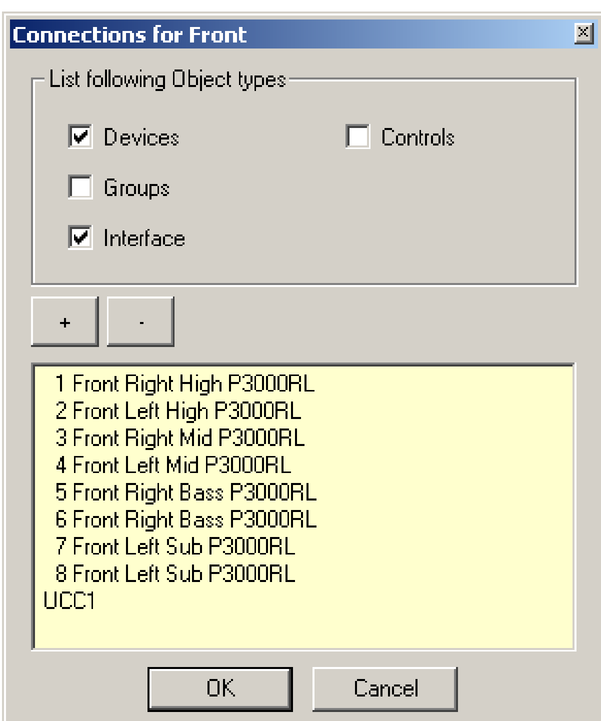
Color	Sets the background color of the group icon in the IRIS-Net worksheet.
Dialog opened by Double Click	Selects the dialog that will be opened when the user double-clicks the group icon in the IRIS-Net worksheet. Which dialogs are available for selection depends on the devices used in the project. Detailed descriptions for the dialogs is provided in the help chapters reference of the related device.
Modify Connections	Opens the Connections dialog to add or remove objects contained in the group.
Close	Closes the Group Class window.
Advanced ...	Opens a window showing extended options for the group.

Connections

The “Connections for...” window is used to select the desired group members.

Objects can be selected in the lower part of the window. A single object can be selected by clicking on it with the left mouse button. Several consecutive objects can be selected by selecting the first object then selecting the last object while pressing the Shift key simultaneously with the mouse click. Individual objects can be toggled between selected and deselected by pressing the Ctrl key simultaneously with the mouse click.

The contents of the list can be determined in two different ways. Either through selecting the types to be shown using the checkboxes in the upper part of the window, or by setting object details using the “+” or the “–“ buttons.



Element	Description
Devices	Allows selection of the Device object-type (devices, for example amplifiers) to be displayed in the object list.

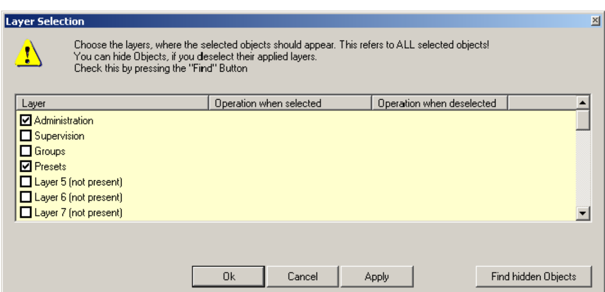
Groups	Allows selection of the Groups object-type (Groups) to be displayed in the object list.
Interface	Allows selection of the Interface object-type to be displayed in the object list.
Controls	Allows selection of the Controls object-type to be displayed in the object list.
+ -	The "+" key increases the amount of detail for the displayed objects, which allows adding individual subparts or sections of an object to the group. The "-" key decreases the amount of detail.
OK	Adds the currently selected objects to the group and closes the “Connections for...” window.
Cancel	Cancels the current object selection and closes the “Connections for...” window.

Editing Object visibility

IRIS-Net offers the possibility to individually specify the visibility status of each object per layer. This option greatly simplifies the design of projects which include layers with similar appearance. Creating an object with its visibility flag unchecked for all layers hides the object from the entire project.

When editing the visibility of an object, please proceed as follows:

1. Highlight the object in the IRIS-Net worksheet
2. Click the right mouse button to open the context menu of the selected object.
3. Select the context menu entry “Edit Object Visibility”. The Layer Selection dialog appears.
4. In the Layer Selection dialog, select the layers on which the previously highlighted object shall be visible.

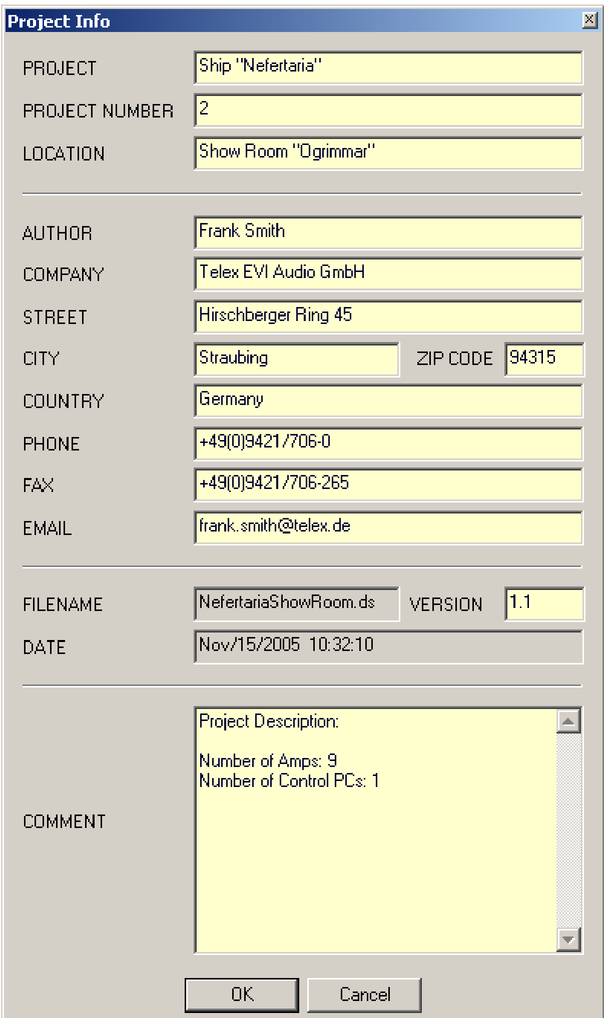


Element	Description
Layer	Layer names within the project
Operation when selected	Operation to be executed when switching to the layer (e.g. RCM24_1*ChA. Mute=1)
Operation when deselected	Operation to be executed when leaving the layer (e.g. RCM24_1*ChA .Mute=1)
Find hidden Objects	Shows all hidden objects on the current layer

How to...

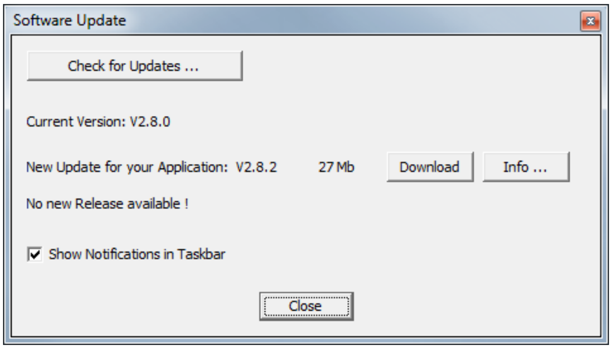
Editing Project Info

The Project Info Dialog provides the user with the possibility to document and save project data in the project file. Among other parameters it is possible to specify project name, project number, location plus additional information on the project author. The COMMENT field provides space for a brief description of the project.
The Project Info Dialog is accessed by selecting the menu entry Info in the Main-Menu.



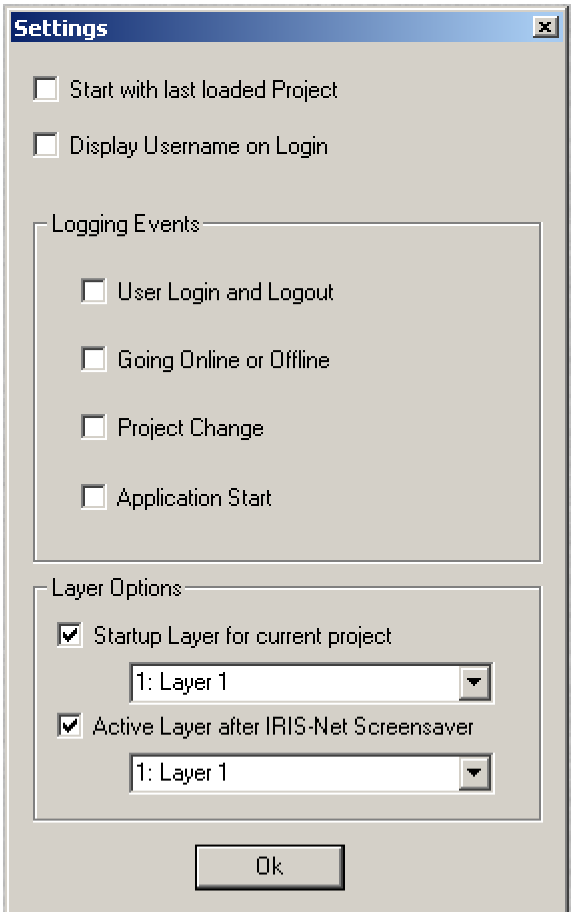
Software update

If a Internet connection is available, updating IRIS-Net is possible via Internet. The Software Update Dialog is accessed by selecting the menu entry ? > Search for Updates ...



Element	Description
Check for Updates..	Checks for software updates at the IRIS-Net update server.
Current Version	Indicates the version of the installed IRIS-Net application.
Download	Press this button to download the update or release.
Info...	Press this button to see information about the update.
Show Notifications in Taskbar	Check this option if notifications about software updates should be shown in the Taskbar.

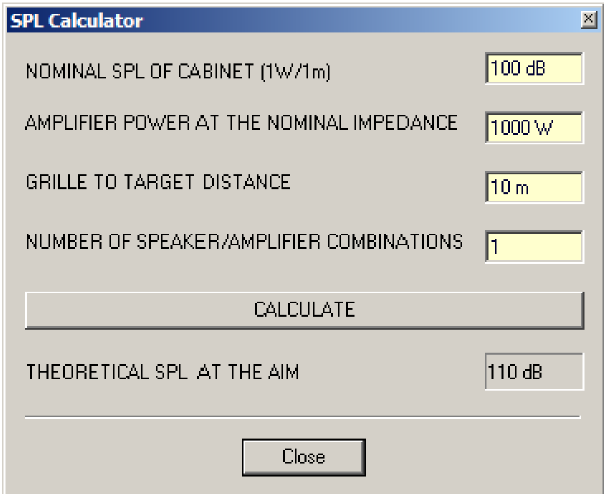
Edit Application Settings
The Settings dialog allows the user to edit attributes of IRIS-Net. Amongst others, the user can specify whether and if so, which events shall be included in the IRIS-Net event log. The Settings dialog is accessed from the menu Edit > Settings....



Element	Description
Start with last loaded Project	Specifies whether the last opened project shall be reopened when IRIS-Net is launched.
Display Username on Login	Specifies whether the user name has to be entered in addition to the password when opening protected project files.
User Login and Logout	Specifies whether user logins and logouts shall be recorded in the event log.
Going Online or Offline	Specifies whether going on-line or off-line shall be recorded in the event log.
Project Change	Specifies whether loading another project shall be recorded in the event log.
Application Start	Specifies whether launching IRIS-Net shall be recorded in the event log.
Startup Layer for current project	Specifies the layer that is shown when the project file is opened.
Active Layer after IRIS-Net Screensaver	Specifies the layer that is shown after the screensaver has been activated.

Using the SPL Calculator

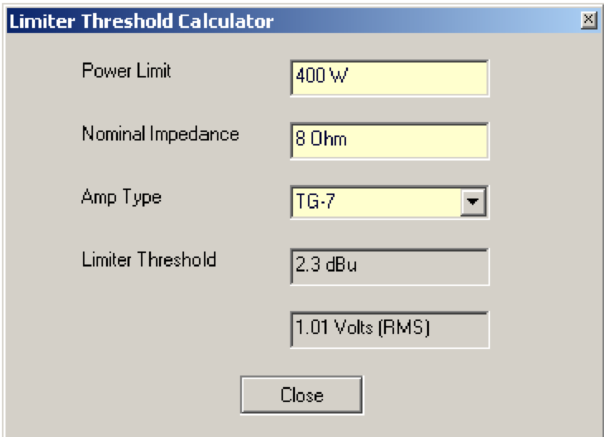
The SPL Calculator provides information about the theoretical SPL resulting from given equipment and given distance between speaker system and sound reinforcement destination. The calculations are based on the theoretically ideal coupling of the single components. Since in practical use several other influences and parameters add to the actually achieved SPL, the results of this computation only represent an approximation.
Selecting ”SPL Calculator” in the Tools Selecting Menu provides access to the SPL Calculator.



Element	**Description**
NOMINAL SPL OF CABINET (1W/1m) 100 dB	This field is for entering the speaker cabinet's average nominal SPL (see data sheet).
AMPLIFIER POWER AT THE NOMINAL IMPEDANCE 1000 W	Enter the power output of the amplifier in this field. Since the amp's actual outputted power depends on the impedance of the load connected, the entered value has to match the load that is actually connected.
GRILLE TO TARGET DISTANCE 10 m	Distance between the cabinet's grille and the aimed destination of the sound reinforcement.
NUMBER OF SPEAKER/AMPLIFIER COMBINATIONS 1	Enter the number of speaker cabinets.
CALCULATE	Clicking this soft key starts the calculation.
THEORETICAL SPL AT THE AIM 110 dB	This field shows the resulting SPL in dB.
Close	Clicking onto this button closes the SPL Calculator.

Limiter Threshold Calculator

The Limiter Threshold Calculator provides information about the limiter threshold to be set for given equipment and given loudspeaker systems data. The Limiter Threshold Calculator makes no claim to be pinpoint accurate; i.e. results should be regarded as benchmarks.
Selecting the entry Limiter Threshold Calculator in the Tools menu provides access to the Limiter Threshold Calculator.



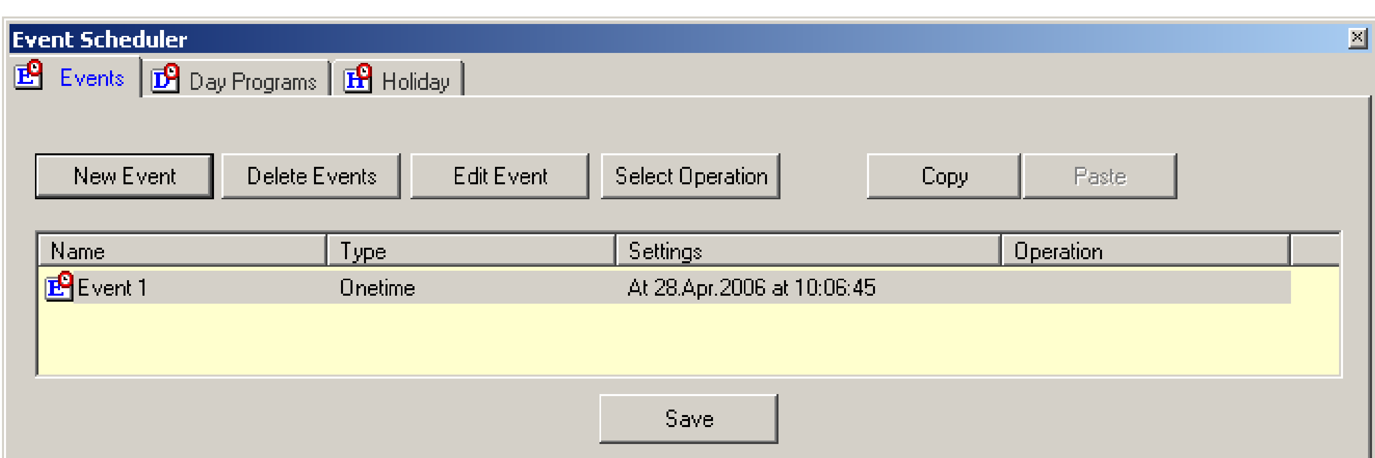
Element	Description
Power Limit	Enter the maximum power consumption of the loudspeaker cabinet used (refer to datasheet) in this field.
Nominal Impedance	Enter the impedance of the loudspeaker cabinet used (refer to datasheet) in this field.
Amp Type	Select the used type of power amplifier in this dropdown menu.
Limiter Threshold	This field shows the result of the calculation in dBu. The value is also shown in dBu.
Close	Clicking onto this button closes the Limiter Threshold Calculator.

Using the Event Scheduler

The Event Scheduler is divided into three windows. On the Events tab you can create and edit nonrecurring as well as periodically recurring events. The Day Programs tab allows creating lists of daily recurring events. The Holiday tab lets you specify periods – like for example school holidays – during which an alternative Day Program shall be used instead of the standard Day Program.

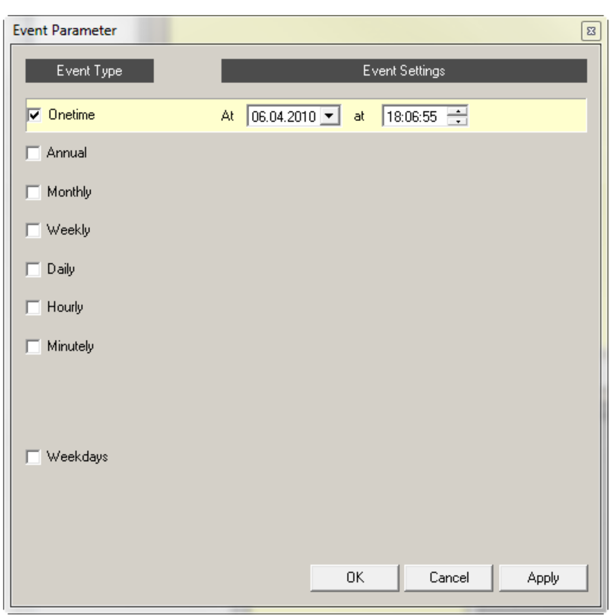
Events

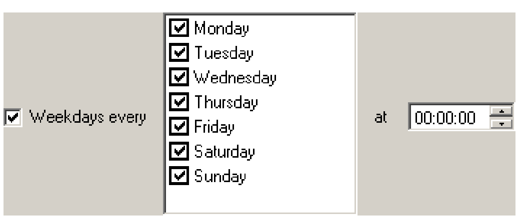
This window allows creating nonrecurring as well as periodically recurring events. If an event occurs, a macro is being executed.



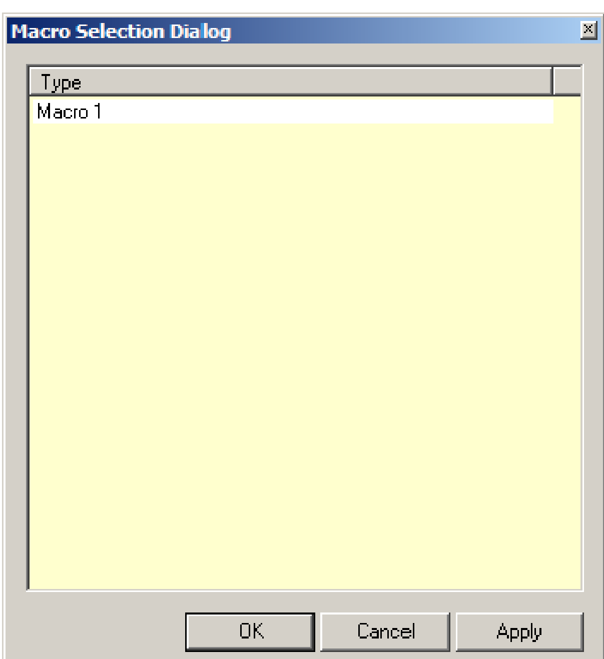
Element	Description
New Event	Creates a new event, which is added at the end of the event list.
Delete Events	Events marked in the Event list will be deleted.
Edit Event	Opens the Event Parameters window for editing the parameters of the selected event.
Select Operation	Opens the Macro Selection Dialog window. Here you are able to select the macro that will be executed when the event occurs that has previously been selected from the event list.
Copy Paste	Copy a single or various marked events from the event list. Copied events can be pasted to the end of the event list.
Name	Event name. Left-click on the name of a previously marked event to enter a new name for this event.
Type	Event type. Double-clicking on an event’s corresponding entry in the event list allows changing the event type in the Event Parameters window.
Settings	Event parameter. Double-clicking on an event’s corresponding entry in the event list allows changing the selected parameter in the Event Parameters window.
Operation	Macro to be executed in case of occurrence of the event. Double-clicking on an event’s corresponding entry in the event list lets you select a macro in the Macro Selection Dialog window.
Save	Saving of modified events in the event list.

Time settings for an event are made in the Event Parameters window. It is possible to program nonrecurring as well as periodically recurring activations of events.



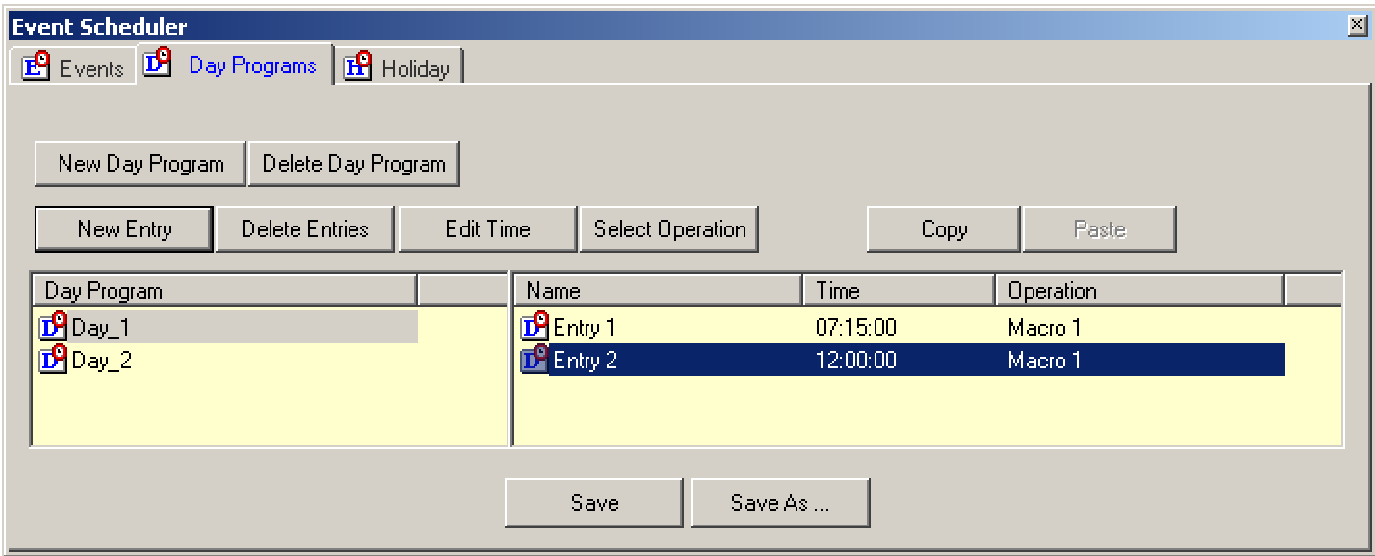
Element	Description
	The event is scheduled to be activated only once on a specific day and at a specific time.
	The event is scheduled to be activated once a year on a specific day of the year and at a specific time.
	The event is scheduled to be monthly activated on a specific day of the month and at a specific time.
	The event is scheduled to be weekly activated on a specific weekday and at a specific time.
	The event is scheduled to be daily activated at a specific time.
	The event is scheduled to be hourly activated at a specific time (min:sec).
	The event is scheduled to be activated every minute at a specific time (seconds).
	The event is scheduled to be activated on selected weekdays of each week and at a specific time.

The Macro Selection Dialog window allows selecting the macro to be executed on the occurrence of an event.



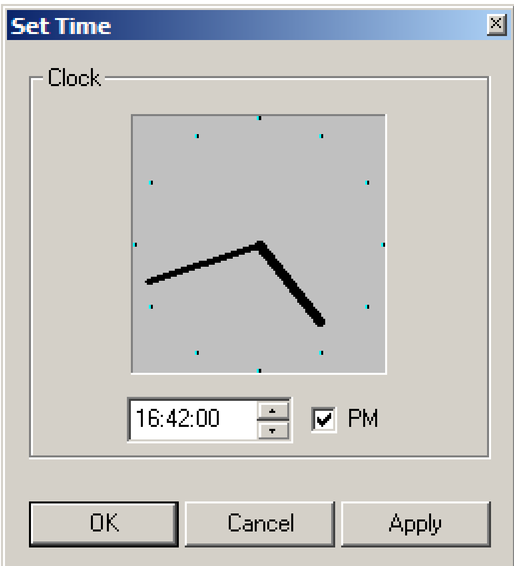
Day Programs

This window allows creating and editing Day Programs. A Day Program represents a list of defined times throughout the day. At each of these points in time a freely selectable macro will be executed.

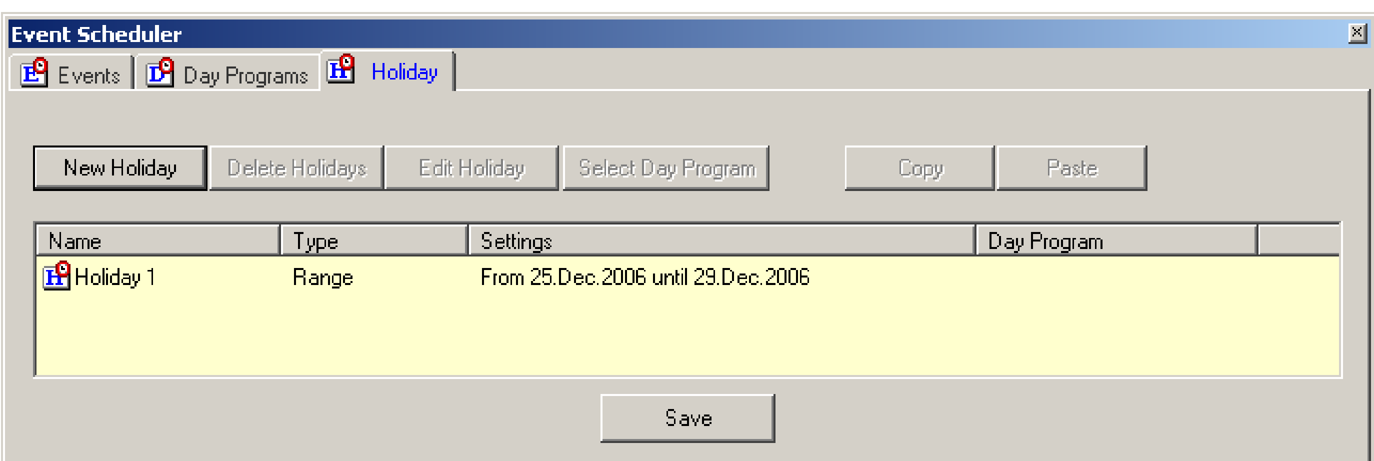


Element	Description
New Day Program	Creates a new Day Program and adds it to the Day Program list.
Delete Day Program	The Day Programs selected in the Day Program list will be deleted.
New Entry	New Entry creates a new entry in the event list of the currently selected Day Program. New events are numbered consecutively with their time being set to 12:00:00 as a standard.
Delete Entries	Events selected in the event list of the Day Program will be deleted.
Edit Time	Opens the Set Time window which lets you edit the time set for the event that is currently selected in the event list.
Select Operation	Opens the Macro Selection Dialog window for selecting the macro to be executed for the event currently selected in the event list.
Copy Paste	Copy a single or various marked events from the event list. Copied events can be pasted to the end of the event list.
Dayprogram	Day Program List. Selecting a Day Program displays a list of associated events which can be edited. Left-click on the name of a pre- viously selected Day Program to enter a new name for this Day Program.
Name	Event name. Left-click on the name of a previously marked event to enter a new name for this event.
Time	Execution time of an event. Double clicking on an event’s corresponding entry in the event list allows setting the time in the Set Time window.
Operation	Macro to be executed on the occurrence of the event. Double-clicking on an event’s corresponding entry in the event list lets you select a macro in the Macro Selection Dialog window.

The Set Time window allows setting the time for an event in the Day Program.
Setting the desired time is possible either by turning the hour hand of the conventionalized clock using the mouse or through entering the numerical value in the time field.



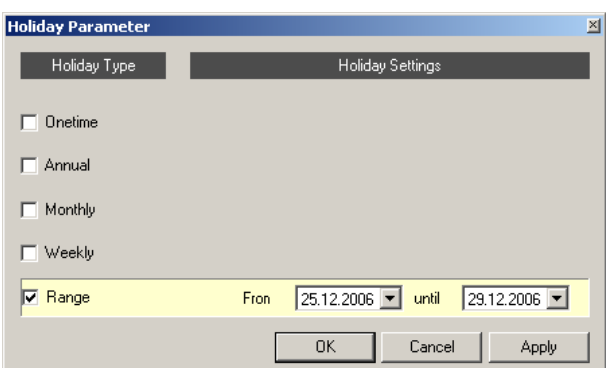
Holiday
This window holds a list of holidays (periods). A specific Day Program is being executed during a holiday. Specifying different holidays offers the possibility to execute various Day Programs during particular periods within a year.



Element	Description
New Holiday	New Holiday creates a new entry in the list of holidays. New holidays are numbered in ascending order, whereas the current date is set as standard (once only) period of a new holiday.
Delete Holidays	The holidays selected in the holiday list are being deleted.
Edit Holiday	Opens the Holiday Parameters window for editing the period of the holiday currently selected in the holiday list.

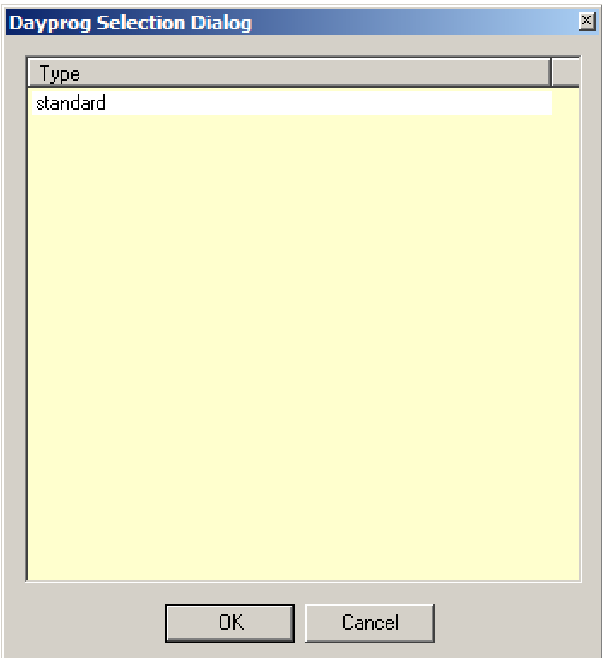
Select Dayprogram	Opens the Dayprog Selection Dialog window for selecting the Day Program to be executed during the period corresponding to the holiday currently selected in the holiday list.
Copy Paste	Copy a single or various marked holidays from the holiday list. Copied holidays can be pasted to the end of the holiday list.
Name	Name of a holiday. Left-click on the name of a previously marked holiday to enter a new name for this holiday.
Type	Type of holiday. Double clicking on the holiday's corresponding entry in the holiday list allows changing the type of that specific holiday in the Holiday Parameters window.
Settings	Parameters of a holiday. Double clicking on the holiday's corresponding entry in the holiday list lets you change parameters in the Holiday Parameters window.
Dayprogram	Day Program to be executed of a holiday. Double clicking on the holiday's corresponding entry in the holiday list allows selecting the Day Program to be executed in the Dayprog Selection Dialog window.
Save	Saving of modified holidays in the holiday list.

The period corresponding to a holiday can be specified in the Holiday Parameters window. A holiday always lasts a single day or several consecutive days. A single day can be one specific date or a periodically recurring date. When selecting several days, dates must be in consecutive order.



Element	Description
Ontime at 06.10.2005 at 09:48:41	The period of a holiday is set to once only for a specific day.
Annual every year at 01.01. at 00:00:00	The period of a holiday is set to once per year for a specific day.
Monthly every month at 1 at 00:00:00	The period of a holiday is set to once per month for a specific day of the month.
Weekly every week at Saturday at 00:00:00	The period of a holiday is set to once per week for a specific weekday.
Range from 24.12.2005 until 02.01.2006	The period of a holiday is set to a time span between two days which need to be specified. These two days are included in the period.

The Dayprog Selection Dialog window allows selecting the Day Program to be executed during a holiday.

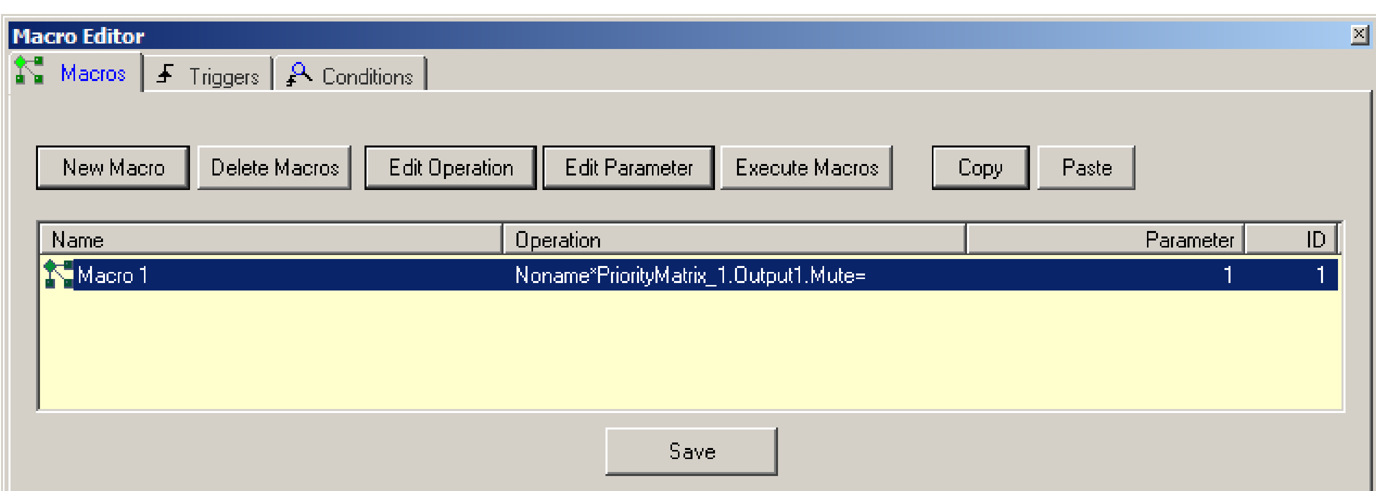


Using the Macro Editor

The Macro Editor window provides access to the Macros tab (see below), Triggers tab and Conditions tab.

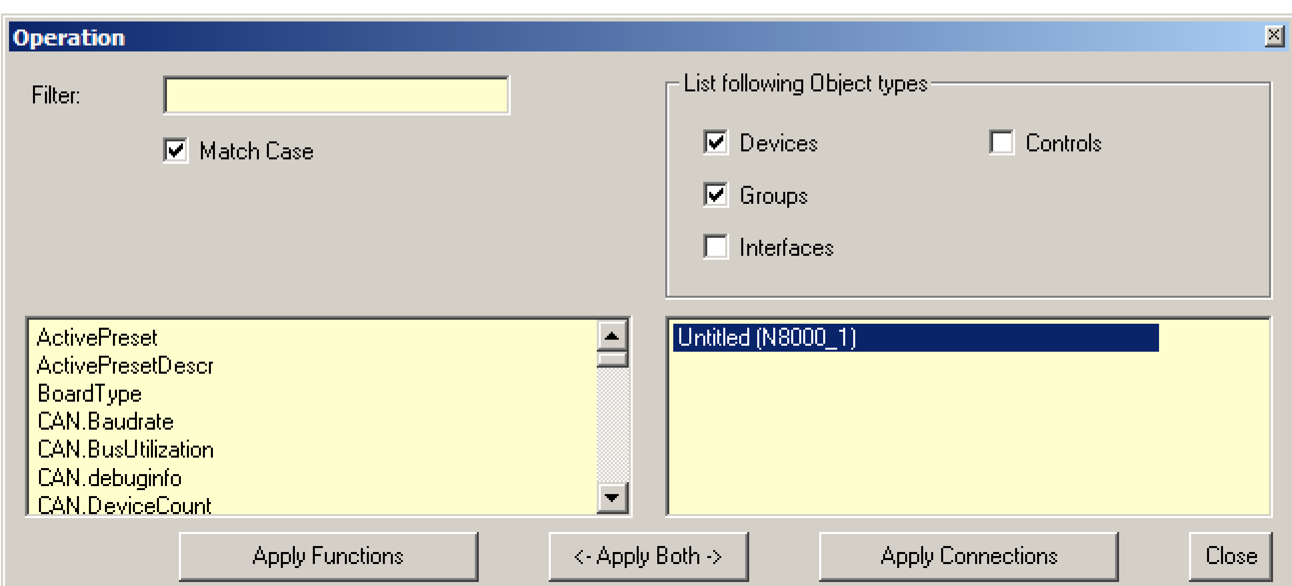
Macros

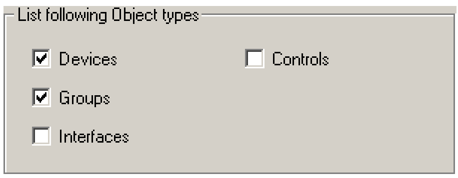
This window allows creating and editing macros. Macros are used to change the status of objects. Objects can be devices (e.g. a N8000 System Controller) as well as elements in IRIS-Net (e.g. controls).



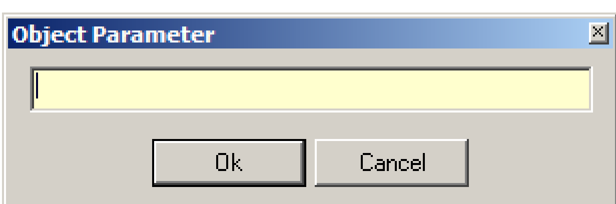
Element	Description
New Macro	Creates a new macro. The macro gets added to the end of the macro list.
Delete Macros	The macros that have been selected in the macro list are deleted.
Edit Operation	Opens the Operation window which allows selecting variables, whose value is to be set by the macro selected in the macro list.
Edit Parameter	Opens the Object Parameter window for setting the value which the macro shall assign to the selected variable.
Execute Macros	Pressing this button executes the macros that have been selected in the macro list.
Copy Paste	Copy a single or various marked macros from the macro list. Copied macros can be pasted to the end of the macro list.
Name	Name of the macro. Left-click on the name of a previously selected macro to enter a new name for this macro.
Operation	The variable that is set when the macro is being executed.
Parameter	The value that is assigned to the variable when the macro is being executed.
ID	System-internal identifier of the macro

The Operation window allows selecting the variable for which the value is to be set by the macro. A variable always consists of an object and the related function. From the window's right section it is possible to choose the desired variable through selecting those types of objects that are to be shown in the object list.
The window's left section allows selecting a function that relates to the previously chosen object.



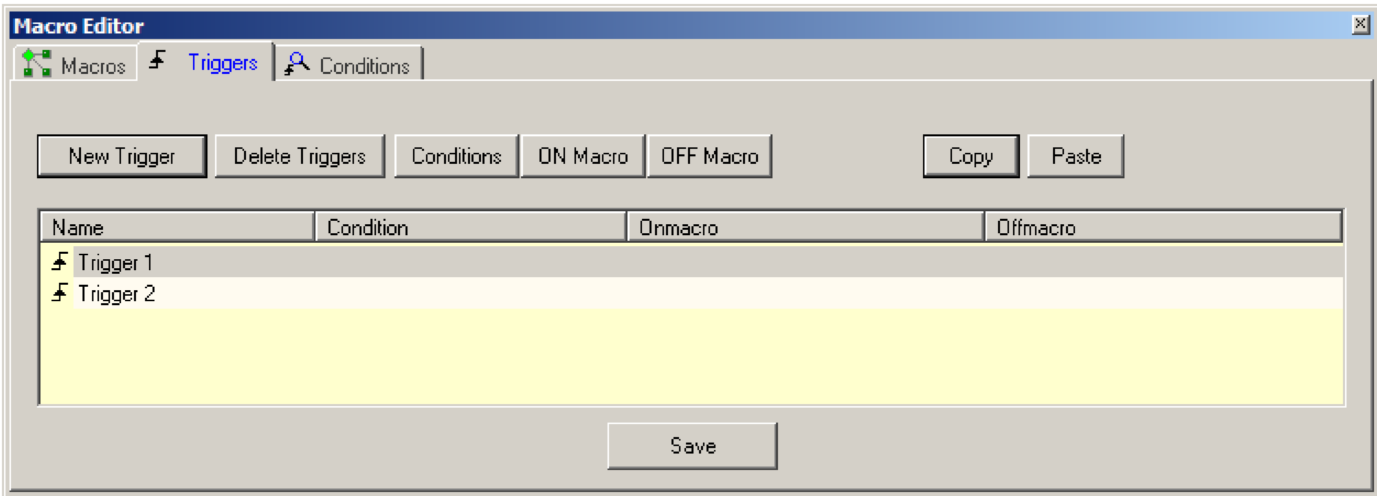
Element	Description
	Only functions that include the text entered here are being displayed in the functions list, which is shown in the left part of the window.
	Check if filter should be case sensitive.
	Selecting a single or various object types is possible. Only objects that comply with the selected types are shown in the list of objects, which greatly simplifies finding the desired object quickly.
	The function that has been selected in the function list is assigned to the macro that is currently being edited.
	The object that has been selected in the object list as well as the function that has been selected in the func- tions list are assigned to the macro that is currently being edited.
	The object that has been selected in the object list is assigned to the macro that is currently being edited.
	Closes the Operation window and returns to the Macros window.

The Object Parameters window allows entering the value that is to be assigned to a variable.



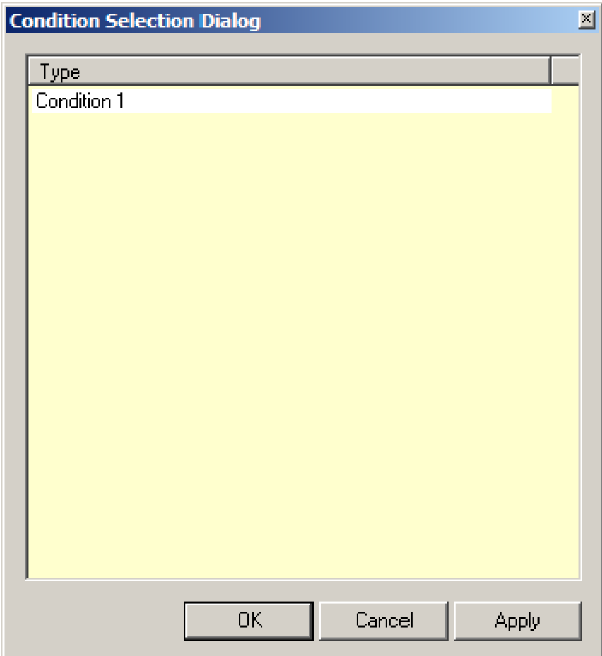
Triggers

Triggers are used to execute macros that depend on the existence of a certain condition. When a condition occurs (when the condition's status changes from „false“ to „true“), a Trigger can start an associated ON-macro. If a condition is not true anymore (when the condition's status changes from "true" to "false"), a Trigger can start an associated OFF-macro.

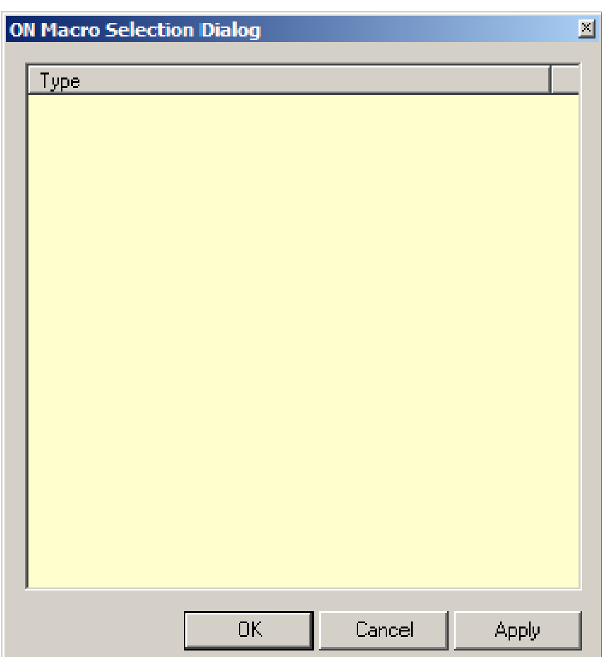


Element	Description
New Trigger	Creates a new Trigger. The Trigger is added at the end of the Trigger List.
Delete Triggers	All Triggers selected in the Trigger List will be deleted.
Conditions	Opens the Condition Selection Dialog Window, which lets the user assign a condition to the Trigger that is currently selected in the Trigger List.
ON Macro	Opens the ON Macro Selection Dialog Window, which lets the user select the macro that is to be executed when the sta- tus of the corresponding condition changes from "false" to "true".
OFF Macro	Opens the OFF Macro Selection Dialog Window, which lets the user select the macro that is to be executed when the sta- tus of the corresponding condition changes from "true" to "false".
Copy Paste	Copies a single or a selection of Triggers previously marked in the Trigger List. The copied Trigger(s) can be pasted at the end of the Trigger List.
Name	The name of the Trigger. Left-clicking on the name of a Trigger that has previously been selected in the Trigger List lets the user assign a new name to the Trigger.
Condition	The condition associated to the Trigger. Double-clicking on the Trigger's corresponding entry in the Trigger List lets the user select the condition to be assigned to the Trigger from the Condition Selection Dialog Window.
Onmacro	The ON-macro of the Trigger. Double-clicking on the Trigger's corresponding entry in the Trigger List lets the user select the macro in the ON Macro Selection Dialog Window.
Offmacro	The OFF-macro of the Trigger. Double-clicking on the Trigger's corresponding entry in the Trigger List lets the user select the macro in the OFF Macro Selection Dialog Window.

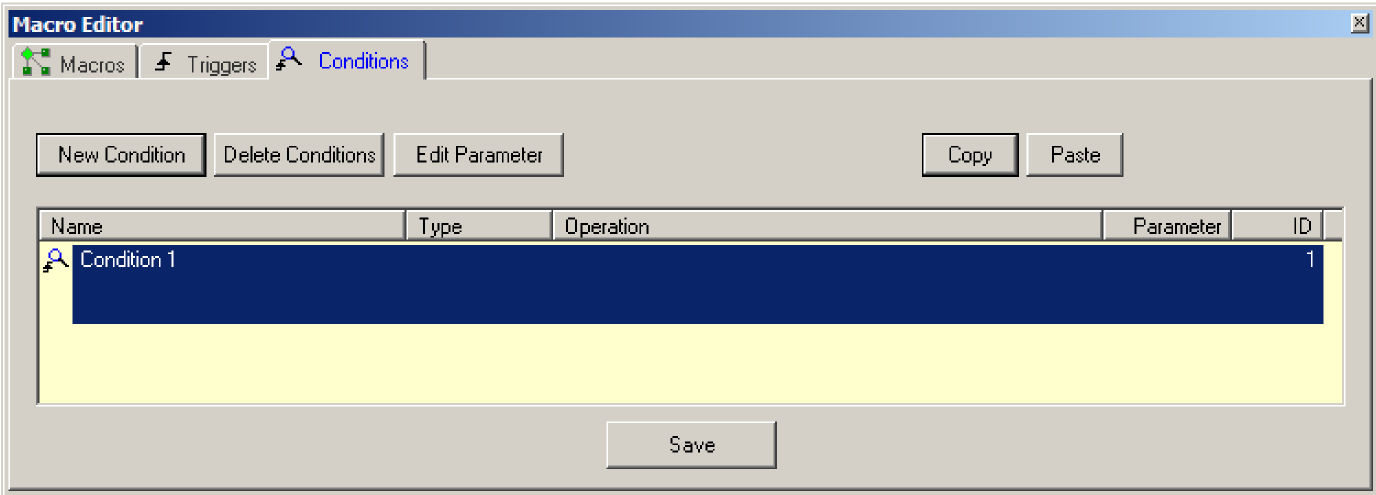
The Condition Selection Dialog Window lets the user select the condition to be associated to the Trigger.



The ON Macro Selection Dialog Window lets the user select the macro that will be executed when the status of the condition changes from "false" to "true". Equally, the OFF Macro Selection Dialog Window lets the user select the macro that will be executed when the status of the condition changes from "true" to "false".



Conditions



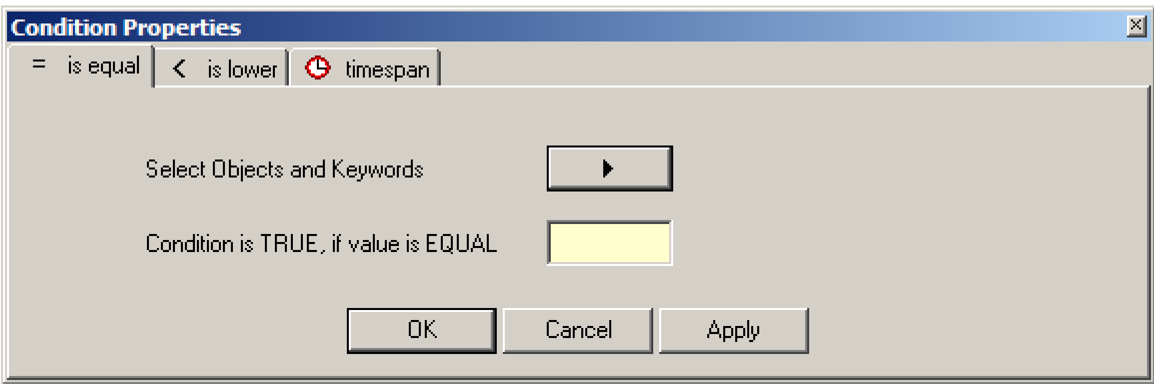
Element	Description
New Condition	Creates a new Condition. The Condition is added at the end of the Condition List.
Delete Conditions	All Conditions selected in the Condition List will be deleted.
Edit Parameter	Opens the Condition Properties Window, which lets the user assign parameters to the Condition.
Copy Paste	Copies a single or a selection of Conditions previously marked in the Condition List. The copied Condition(s) can be pasted at the end of the Condition List.
Name	The name of the Condition. Left-clicking on the name of a Condition that has previously been selected in the Condition List lets the user assign a new name to the Condition. The magnifier symbol in front of a Condition's name signals the current status of the Condition. A white magnifier symbol indicates that the Condition is "false" while a green magnifier symbol indicates that the Condition is "true".
Type	The type of a Condition can be "is equal", "is lower" or "timespan". Double-clicking on a Condition's corresponding entry in the Condition List lets the user select the type for the Condition in the Condition Properties Window.
Operation	The properties of a Condition. Double-clicking on a Condition's corresponding entry in the Condition List lets the user select the properties of a Condition in the Condition Properties Window.
Parameter	The value that the selected variable has to equal or fall below.
ID	System internal identifier of a Condition.

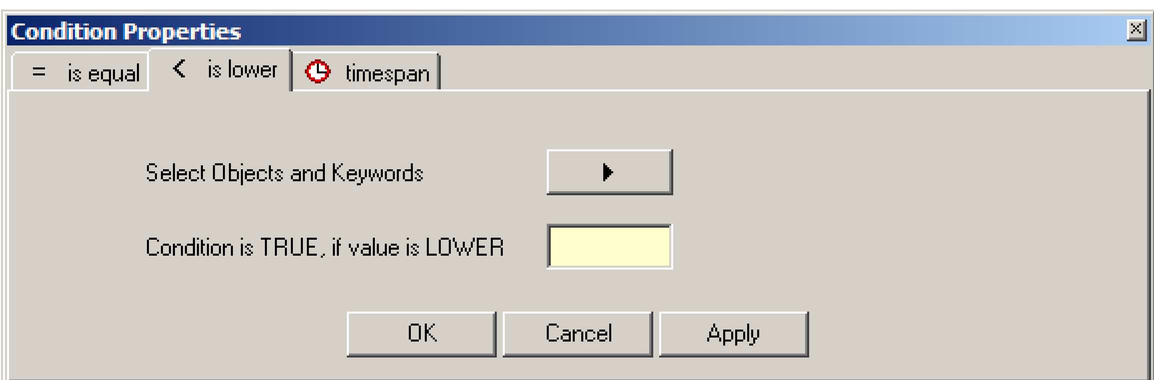
The Condition Properties Window lets the user configure a Condition. The configuration determines whether a Condition is "true" or "false". There are three different types of conditions that constitute a configuration:

– "Is equal": a condition is true when a variable exactly equals a specific value. Otherwise, the condition is false.

– "Is lower": a condition is true when a variable is truly lower than a specific value. Otherwise, the condition is false.
– "Timespan": within a certain period of time the condition is true either exactly once or periodically (in adjustable intervals) for a specified time span. At all other times the condition is false.

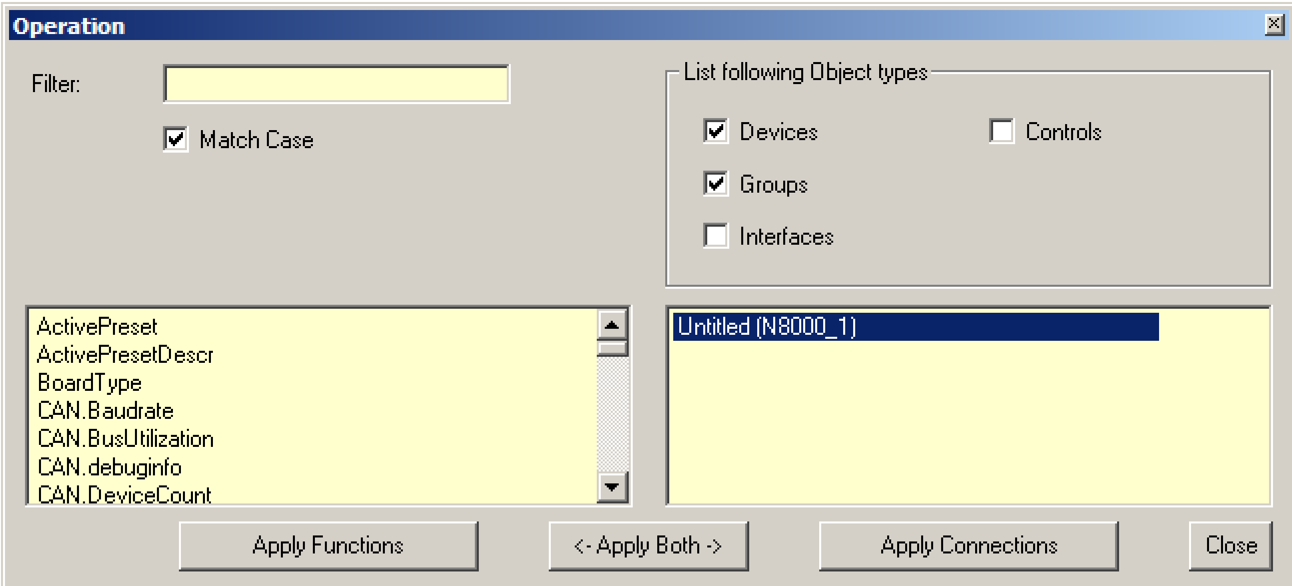
Condition in dependency of a variable
The Dialog for the configuration of a 'Condition in dependency of a variable' is identical structured for the logical operators "is equal" and "is lower".





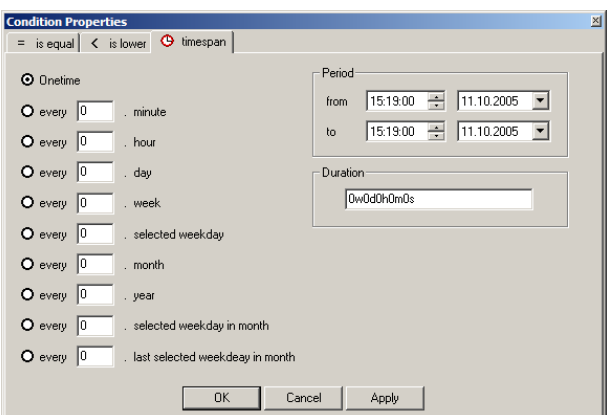
Element	Description
Select Objects and Keywords	Opens the Operation Window, which lets the user select the variable (= objects and keywords) that is to be used for the equation/comparison.
Condition is TRUE, if value is LOWER	The value that the selected variable has to equal or fall below.

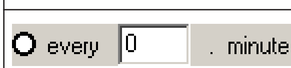
The Operation Window lets the user select the variable whose value is relevant for the Condition. A variable always consists of an object and an associated function. To select a variable, the window's right frame lets the user select the types of objects that will appear in the Object List. Now, the function to be associated to the previously selected object can be selected in the left frame of the window.

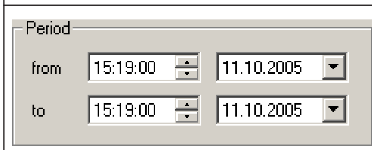
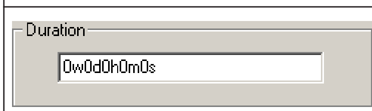
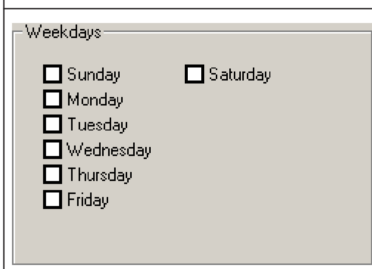


Timed Condition

A Timed Condition is always defined by a Period of Time, during which the Condition is "true" for a specified Duration. The Period of Time may occur once (Onetime) or periodically (every ...). Appropriately, the Duration should not exceed the selected Period of Time.

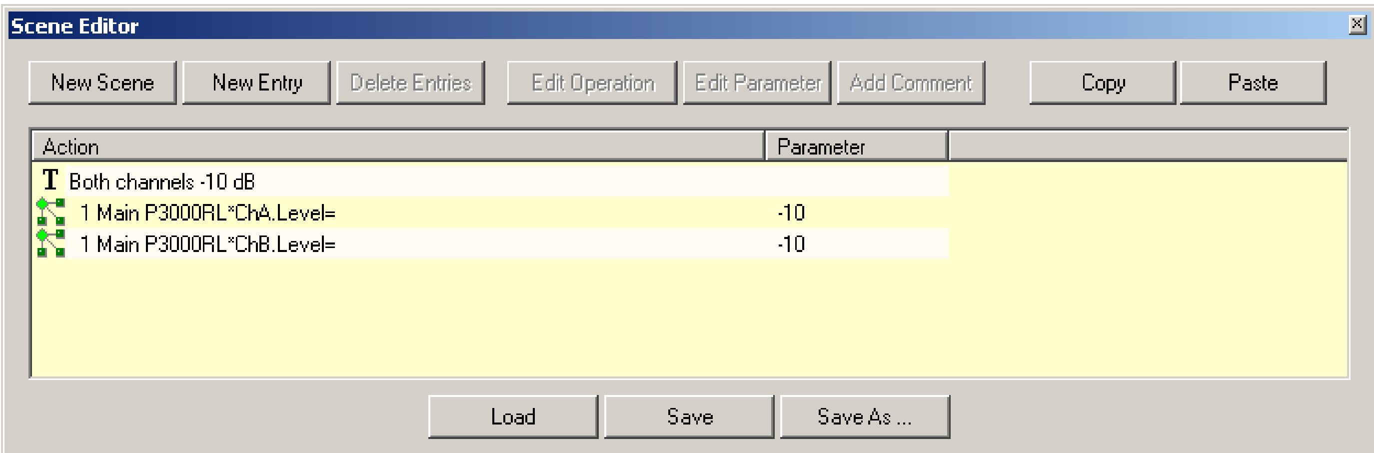


Element	Description
	Starting from the beginning of the Period of Time, the Condition is “true” exactly once for the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x minute(s) during the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x hour(s) during the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x day(s) during the Duration.

	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x week(s) during the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is "true" for the Duration, if the first day of the Period of Time falls on the selected weekday. Otherwise, starting from the beginning of the Period of Time, the Condition is "true" on every selected x weekday(s) at start time of the Period of Time and for the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x month(s) during the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is recurringly "true" every x year(s) during the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is "true" for the Duration, if the first day of the Period of Time falls on the selected x weekday of a month. Otherwise, starting from the beginning of the Period of Time, the Condition is "true" on every selected x weekday(s) of a month at start time of the Period of Time and for the Duration.
	Starting immediately from the beginning of the Period of Time, the Condition is "true" for the Duration, if the first day of the Period of Time falls on the selected x weekday of a month. Otherwise, starting from the beginning of the Period of Time, the Condition is "true" on every selected x last weekday(s) of a month at start time of the Period of Time and for the Duration.
	The Period of Time has a start time (start time on a start day) and an end time (end time on an end day). Within this Period of Time the Condition is once or periodically "true" for the Duration.
	The Duration during which the Condition shall be "true" at one point in time or at numerous points in time. The syntax for entering the Duration is: week (w), days (d), hours (h), minutes (m), and seconds (s). Values have to be separated by the letters indicated in parentheses. Example: A Condition shall be "true" for a time span of 90 minutes. Some of the allowable syntaxes for entering the Duration are: 0w0d1h30m0s; 1h30m; 90m; 89m60s
	For a Condition that is periodically "true" on specific weekdays, the desired weekday(s) can be selected. The List of Weekdays is displayed only if one of the following types of Conditions has been activated: selected weekday, selected weekday in month or last selected weekday in month

Using the Scene Editor

A Scene consists of a freely configurable number of actions. Examples for actions that can be part of a Scene are: DSP parameters, GPIO settings, remote amplifier parameters or the loading of presets.



Element	Description
New Scene	This creates a new Scene. Creating a new Scene closes the currently opened Scene. A window opens requesting the user to acknowledge this step: "Do you want to save your changes?" Button Yes: Saves the currently opened Scene. In case the currently opened Scene already has a name and has previously been saved, the Scene is simply overwritten. The “Save as...” dialog window opens when a Scene has not been saved before. This window allows assigning a name to a Scene. Button No: Changes applied to a Scene are not being saved. Button Cancel: Cancels the creation of a new Scene.
New Entry	Creates a new entry in a Scene.
Delete Entries	Deletes the currently marked entries of a Scene.
Edit Operation	Lets the user edit the column-entry Action of the selected Scene entry by opening the Operation dialog.
Edit Parameter	Lets the user edit the column-entry Parameter of the selected Scene entry by opening the Object Parameter dialog.
Add Comment	Adds a comment line to a Scene.
Copy Paste	Copying or Pasting in a single or various Scene entries.
Action	Action that is being executed when recalling a Scene.
Parameter	Parameter of an action that is being executed when recalling a Scene.
Load	Loads a previously saved Scene.

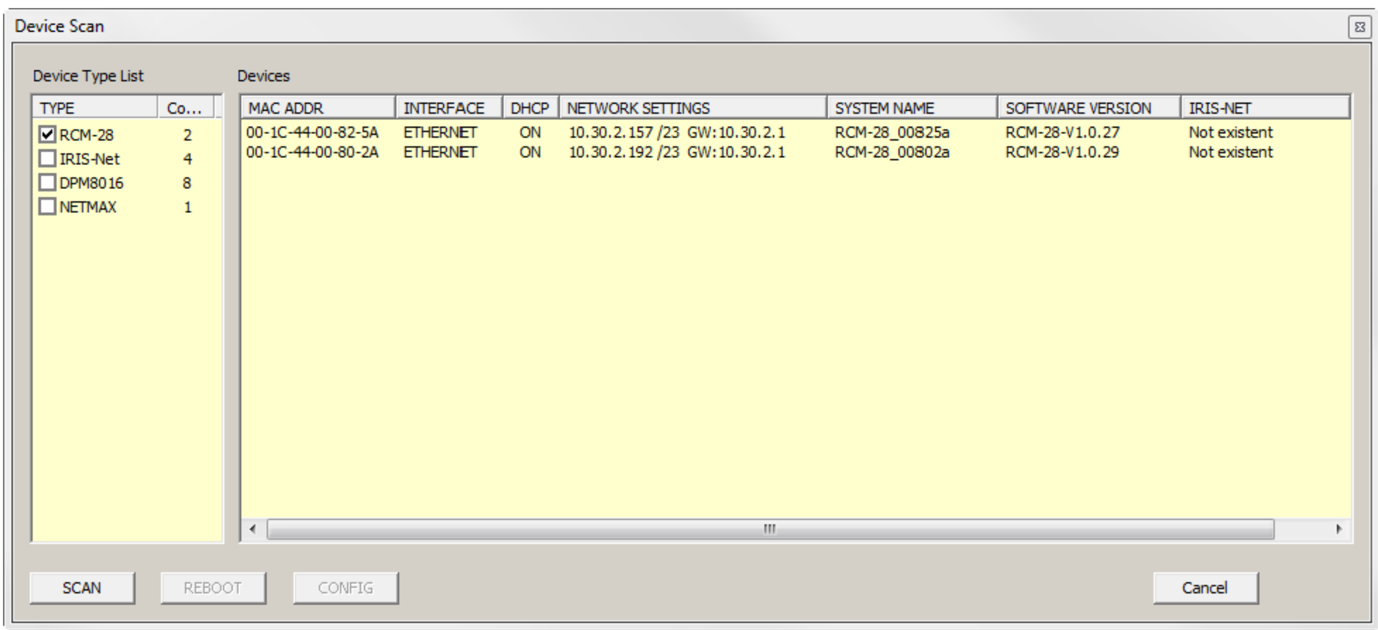
Save	Saves a Scene.
Save As ...	Saves the Scene under a different name.

Searching/Configuring devices on the Ethernet

The Device Scan dialog allows configuration of the network interfaces of non-OMNEO-devices that are accessible from the PC via Ethernet. A device can be configured even where illegal/incorrect network settings would prevent normal communication (i.e., an invalid or conflicting IP address). Use the OCA Scan dialog (see *Searching OMNEO-devices on the Ethernet, page 66*) to scan for OMNEO devices.

HINT: Firewalls can cause problems when using the IRIS-Net Device Scan.

Selecting the entry Device Scan in the Tools menu lets you access the Device Scan dialog.



Element	Description
Device Type List	Shows the types of devices which, upon pressing the SCAN button, are accessible via Ethernet. Selecting a list entry lists only devices of this specific type.
MAC ADDR	Shows the MAC address of the detected devices
INTERFACE	Shows the interface of the detected devices.
DHCP	Shows the status of the DHCP parameter of the detected devices. If DHCP is "ON" the network settings for a device are assigned by a DHCP ser- ver. If DHCP is "OFF" network settings must be configured manually. **HINT: Activate DHCP (“ON”) only if a DHCP server is available in your Ethernet.**
NETWORK SETTINGS	Shows the network settings of the device in the format: IP address / subnet mask GW: gateway address
SYSTEM NAME	Shows the system name of the device

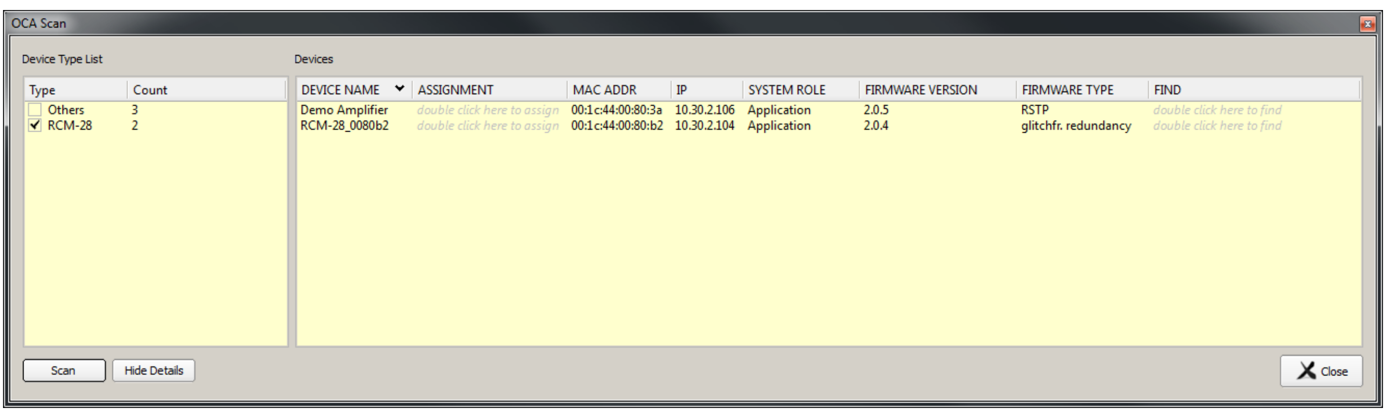
SOFTWARE VERSION	Shows the firmware version of the device
IRIS-Net	If a device is part of the current project, this column shows the name assigned to the device in this specific project.
SCAN	Starts a search for devices on the Ethernet network.
REBOOT	Use this button to restart the device that has been selected in the devices list. Before the device restarts, a dialog box appears, asking the user to enter user name and password.
CONFIG	Use this button to configure the device that has been selected in the devices list. Pressing the CONFIG button opens the Config dialog box.
Cancel	Clicking onto this button closes the Device Scan dialog.

Searching OMNEO-devices on the Ethernet

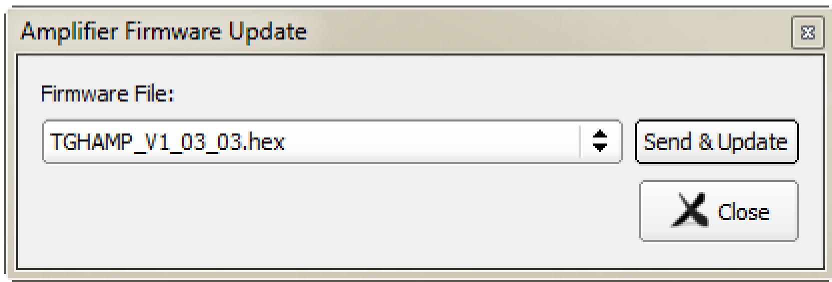
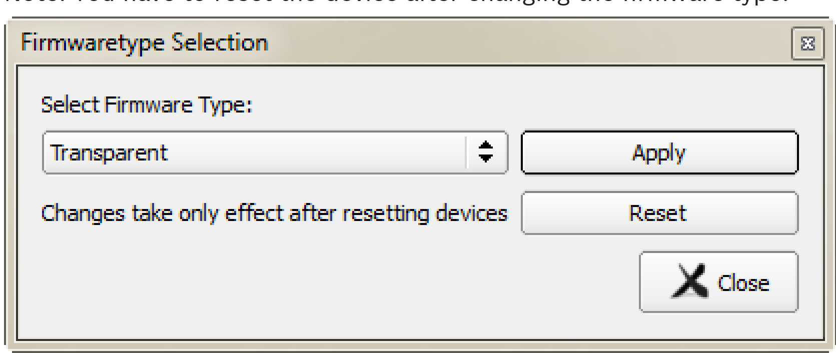
The OCA Scan dialog allows configuration of OMNEO-devices that are accessible from the PC via Ethernet.

HINT: Firewalls can cause problems when using the OCA Scan dialog.

Selecting the entry OCA Scan in the Tools menu lets you access the OCA Scan dialog.

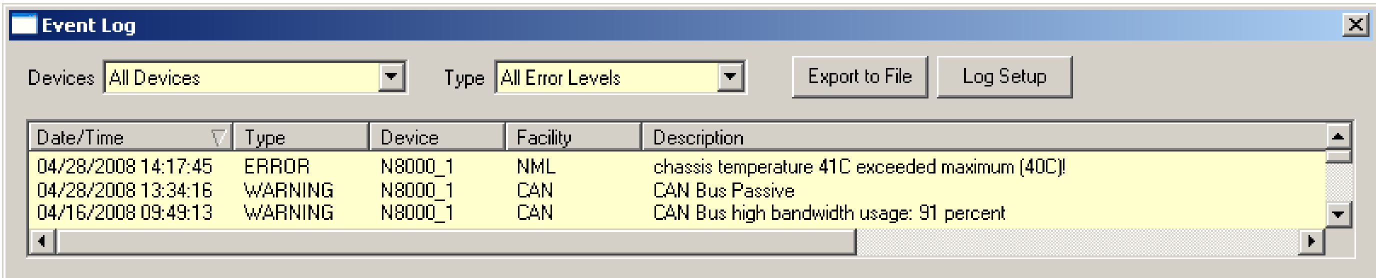


Element	Description
Device Type List	Shows the types of devices which, upon pressing the SCAN button, are accessible via Ethernet. Selecting a list entry lists only devices of this specific type.
Count	Number of devices accessible via Ethernet.
DEVICE NAME	Shows the MAC address of the detected devices. **HINT: There are two drag & drop functionalities for devices listed here. The first option is to drop a device from the OCA Scan dialog into the IRIS-Net worksheet. A new RCM-28 remote amplifier will be created and automatically linked to the dropped device. The second option is to drop a device from the OCA Scan dialog onto an existing RCM-28 remote amplifier in the worksheet. By doing this the existing RCM-28 remote amplifier will be renamed and linked to the dropped device.**
IRIS-NET NAME	Double click to assign a IRIS-Net device name to the OMNEO device.

<table>
<tr><td>MAC ADDR</td><td>Shows the MAC address of the detected devices.</td></tr>
<tr><td>IP</td><td>Shows the IP address of the detected devices.</td></tr>
<tr><td>SYSTEM ROLE</td><td>Shows the system role of the device.</td></tr>
<tr><td>FIRMWARE VERSION</td><td>Shows the firmware version of the device.
To update the amplifier firmware, select one or more amplifiers in the devices list, then right click in this column. The amplifier firmware update dialog appears, there you can select the new firmware version and initiate the update procedure by clicking on the Send & Update button.

</td></tr>
<tr><td>FIRMWARE TYPE</td><td>Shows the firmware type of the device.
To edit the firmware type, select one or more amplifiers in the devices list, then right click in this column. The Firmwaretype selection dialog appears, there you can select the firmware type depending on your network topology or application:
– Transparent
– RapidSpanningTree
– GlitchFreeAudio
Note: You have to reset the device after changing the firmware type.

</td></tr>
<tr><td>FIND</td><td>Double click to activate the find function of the OMNEO device.</td></tr>
<tr><td>Scan</td><td>This button starts a search for devices on the Ethernet network.</td></tr>
<tr><td>Close</td><td>Clicking onto this button closes the OCA Scan dialog.</td></tr>
</table>

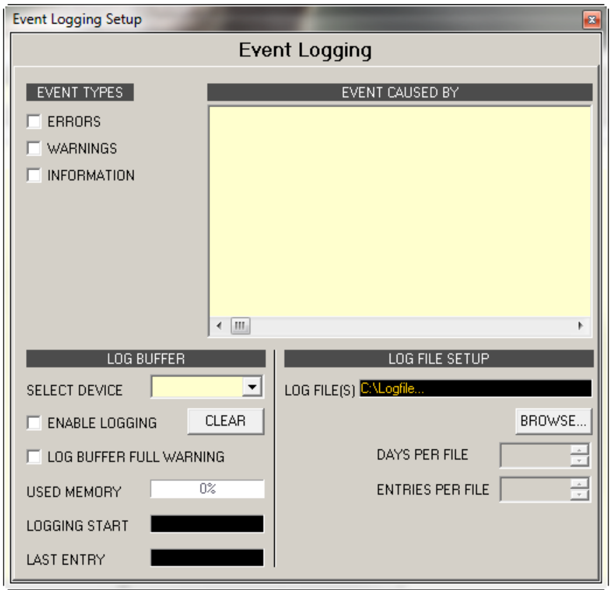
Using the PA Event Log

The Event Log chronologically lists internal IRIS-Net events and events that occurred during the operation of devices that are included in the project. This, for example, aids when troubleshooting the system.



Element	**Description**
Devices	Allows the selection of devices or device types to be included in the Event List.
Type	Allows the selection of fault types to be included in the Event List.
Export to File or Download Logs from Selected Devices	Entries selected in the Event List are saved in a file.
Log Setup	Opens the Event Logging Setup dialog.
Date/Time	Date and time of an event.
Type	The type of an event.
Device	The device that triggered the event.
Facility	The sub-system of the device that triggered the event.
Description	Textual description of an event.

Event Logging Setup
This dialog allows specifying the event type to appear in the Log File. Events are defined as problems (errors) that occurred within the system, but also messages providing information about the system status or changes in the status. Selecting event types is possible in two different ways: either by selecting the TYPE of an event or by selecting sub-systems of the devices employed in the project to be monitored. In addition, this dialog offers the possibility to edit the logging characteristics of N8000s or DPM 8016s that are included in the project and the characteristics of the central logging file that is stored on the PC.

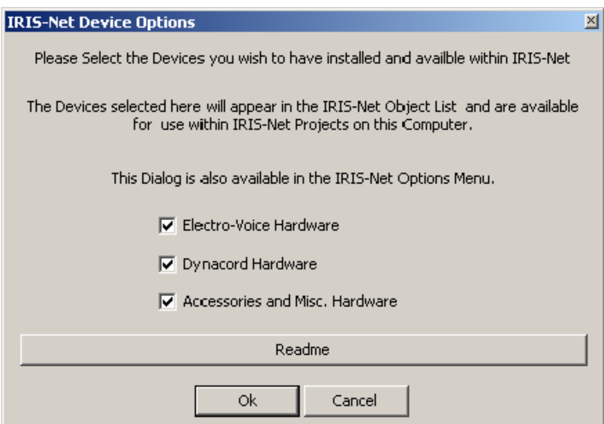


Element	Description
EVENT TYPES – ERRORS – WARNINGS – INFORMATION (N8000/P 64) or AUDIOEVENTS (DPM 8016)	Selection of event types to appear in the Event Log.
EVENT CAUSED BY	Selection of devices or sub-systems to appear in the Event Log.
SELECT DEVICE	Selection of a N8000 or DPM 8016 in the project whose logging characteristics are to be edited.
ENABLE LOGGING (N8000/P 64 only)	The selected device writes an Event Log when this checkbox is checked.
CLEAR	Deletes the entire Event Log of the selected N8000/P 64 or DPM 8016.
LOG BUFFER FULL WARNING (N8000/P 64 only)	An entry appears in the Event Log if this checkbox is checked and the Event Log memory of the device runs low.
USED MEMORY	Displays the available Event Log memory of a the device. 0% indicates that the memory is empty.
LOGGING START (DPM 8016 only)	Displays date and time of the first (e.g. oldest) entry in the Event Log.
LAST ENTRY (DPM 8016 only)	Displays date and time of the last (e.g. newest) entry in the Event Log.
LOG FILE(S) (N8000/ P 64 only)	Displays the storage location at which the currently used log file is saved on the PC in relation to the IRIS-Net installation path.

BROWSE... (N8000/ P 64 only)	Opens a dialog for selecting the log file.
DAYS PER FILE (N8000/ P 64 only)	Number of days after which a new log file is being created.
ENTRIES PER FILE (N8000/P 64 only)	Number of entries at which, when exceeded, a new log file is being created.

Changing available devices

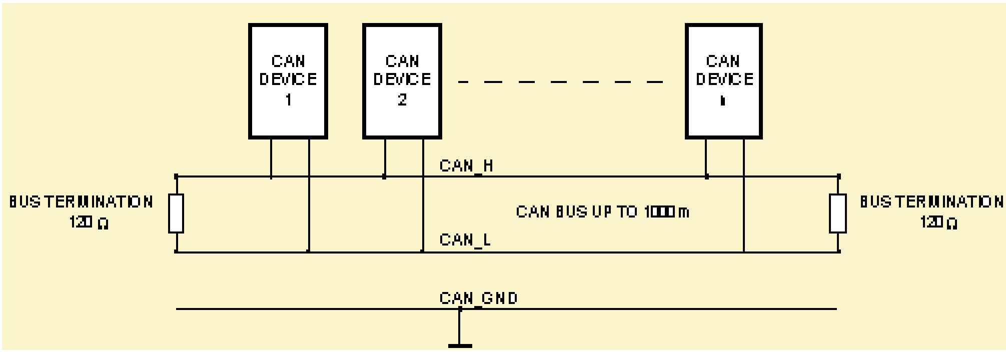
The devices shown in the Object List can be customized. Use the IRIS-Net Device Options dialog window. The dialog window opens automatically the first time IRIS-Net is run. For later customization, you can find the dialog window in the menu Edit > Device Options.



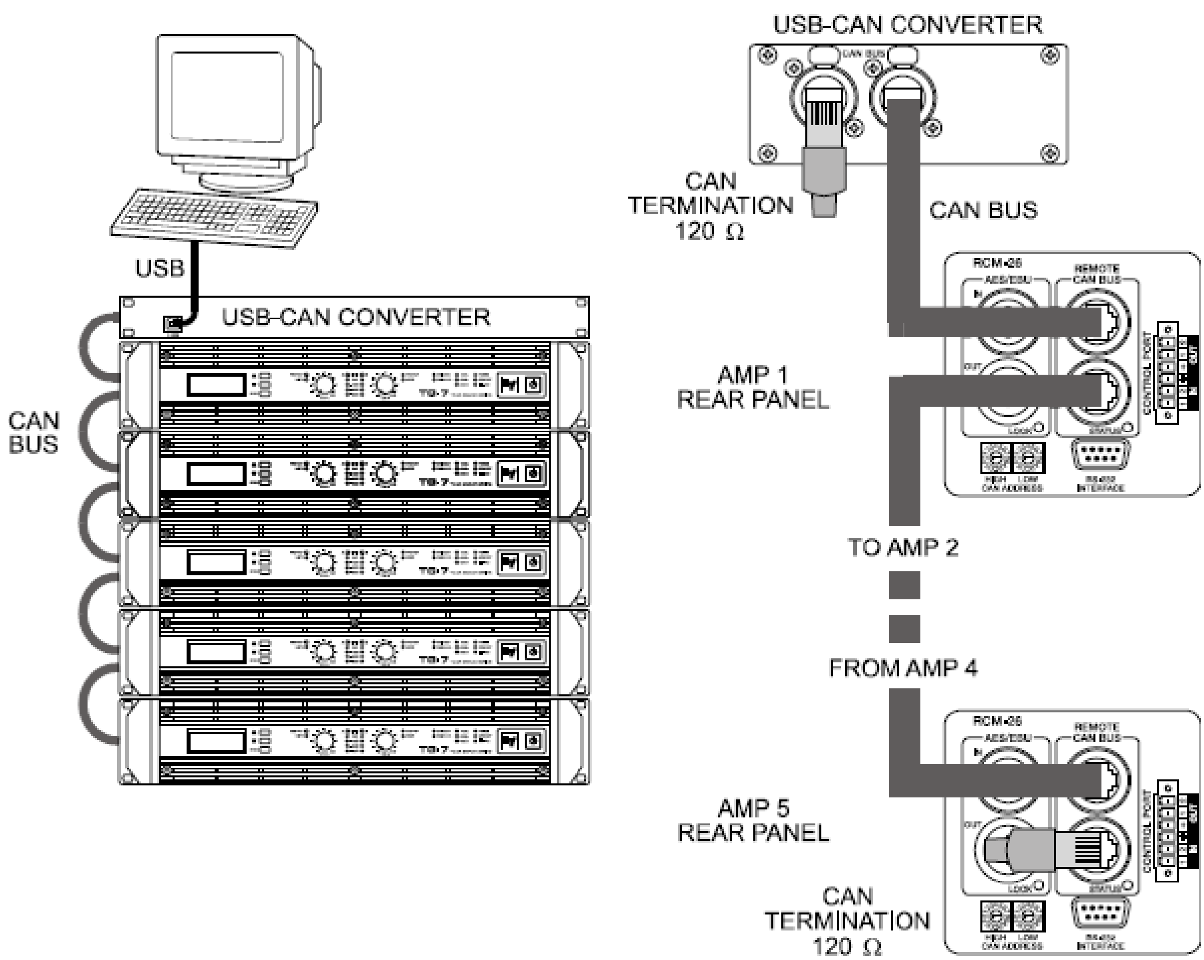
Select the groups of devices that should be available in the Object List. Project files containing devices that are not selected in this dialog can still be opened and edited.

Remote Control Network / Interface

The network for the Remote Control power amps is based on the CAN-bus standard. This popular protocol has been used for many years in automotive, industrial and security applications. The CAN-bus is a balanced serial interface for command and data transmission. Controlling the power amplifiers is performed from a Windows based PC running IRIS-Net – Intelligent Remote & Integrated Supervision – software. The UCC1 USB-CAN Converter serves as the interface between the PC and the CAN-bus. For more detailed information please refer to the UCC1 owner's manual. Up to 100 devices can be connected to a single CAN-Bus with a maximum total cable length of 1,000 metres. An additional CAN-bus is needed for controlling more than 100 devices while the IRIS-Net software will support a total of 250 amplifiers. The network topology used by the CAN-bus is based on a "bus or line topology", i.e. all participants are connected via a single two-wire cable (Twisted-Pair cable, shielded or unshielded) with the cabling daisy-chained from one participant on the bus to the next, allowing unlimited communication among all devices. In general, it does not matter whether a participant on the bus is a power amplifier or a UCC1 USB-CAN converter, this flexibility allows a UCC1 (and its associated PC) to be inserted at any position on the network. Incorporating several UCC1's on a single CAN-bus is also possible. A total of up to 100 devices can be operated on a single CAN-bus. Since the CAN-interfaces of all EV/DC devices are galvanically isolated from the rest of the circuitry, network cabling also carries a common ground conductor (CAN_GND) ensuring that all CAN-interfaces in the network are connected to a common ground potential. The UCC1 provides the possibility for switching the CAN-ground to circuit-ground.



Each participant on the bus system has two RJ-45 connectors for the Remote CAN-bus. These sockets are connected in parallel to serve as input and output (for connecting through) for the data transfer within the remote-network. The CAN-bus must be terminated at both ends using 120 Ω terminator plugs, two of which – CAN-TERM 120 Ω – are supplied with the UCC1. Connect one of these to the RJ-45 socket of the first and the other to the socket of the last appliance on the CAN-bus.
The following illustration shows an example of the data-bus wiring.



In addition to the CAN-bus data signal, network cabling also carries the balanced monitor audio signal for monitoring the power amp inputs and outputs. This monitor-bus allows software-controlled monitoring of the input and output signals of all power amps that are included in the remote network, without the need for additional wiring. This balanced line level audio signal is present at the UCC1’s XLR-type MONITOR Output connector. Typical uses include connecting

it to a spare input of a mixing console or an active monitor speaker so that an engineer can easily monitor the audio signal at the input or output of any amplifier on the network. The CAN-bus standard provides several different data transfer rates, with the data rate being indirectly proportional to the bus cable length. Small networks allow baud rates up to 500 kbit/s. For very large networks reducing the baud rate (minimum 10 kbit/s) is necessary. The following table indicates the relation between baud rate and bus length or in other words network dimensioning:

Data transfe rrate	**Bus Length**
500 kbit/s	100 m
250 kbit/s	250 m
125 kbit/s	500 m
62,5 kbit/s	1000 m
20 kbit/s	2500 m
10 kbit/s (Default)	5000 m

With all remote power amps, the factory data rate setting defaults to 10 kbit/s. The use of repeaters is generally recommended when the bus-length exceeds 1,000 m.

CAN-Bus Cable Specifications

According to the ISO 11898-2 standard, CAN-bus data transfer cabling has to be carried out using Twisted-Pair cables with or without shielding providing a characteristic impedance of 120 Ω. Both ends of a CAN-bus need to be terminated with 120 Ω termination-plugs.

The maximum bus-length depends on the actual data transfer rate, the kind of data transfer cable being used, and the total number of participants on the bus. The following table shows the most essential parameters for CAN-networks consisting of up to 64 participants:

Bus Length (inm)	**Data Transmission Cable**		**Termination (inΩ)**	**Max. Data Transfer Rate**
	Resistance per Unit Length (in mΩ/m)	**Cable Diameter**		
0...40	< 70	0,25...0,34 mm² AWG23, AWG22	124	1000 kbit/s bei 40 m
40...300	< 60	0,34...0,6 mm² AWG22, AWG20	127	500 kbit/s bei 100 m
300...600	< 40	0,5...0,6 mm² AWG 20	150...300*	100 kbit/s bei 500 m
600...1000	< 26	0,75...0,8 mm² AWG 18	150...300*	62,5 kbit/s bei 1000 m

* With longer cables and many participants on the CAN-bus, termination resistors with higher impedance than the specified 120 Ω are recommended to reduce the ohmic load of the interface drivers and therefore the voltage drop between the two cable-ends.

The following table is meant for first assessment of necessary cable diameters for different bus lengths and bus-participant numbers:

Bus Length (inm)	Number of Units on the CAN-Bus		
	32	**64**	**100**
100	0,25 mm² or AWG24	0,34 mm² or AWG22	0,34 mm² or AWG22
250	0,34 mm² or AWG22	0,5 mm² or AWG20	0,5 mm² or AWG20
500	0,75 mm² or AWG18	0,75 mm² or AWG18	1,0 mm² or AWG17

Additionally, the length of branch lines – for participants that are not directly connected to the CAN-bus – is also of importance. For data transfer rates of up to 125 kbit/s, the maximum length of a single stub cable should not exceed 2 m. For higher bit rates a maximum length of only 0.3 m is recommended. The entire length of all branch lines in a network should not exceed 30 m.

General Note:

– As long as only short distances (up to 10 m) are concerned, common RJ-45 patch cables with 100 Ω characteristic impedance (AWG 24 / AWG 26) can be used for the cabling inside of a rack-shelf system.
– The previously outlined guidelines for network cabling are mandatory as far as the rack-shelve interconnection or fixed installations are involved.

Setting up a Remote Amplifier System

A Remote Power Amplifier System is a computer-controlled audio system that consists of a single or multiple Remote Power Amps and a single or multiple PCs with the IRIS-Net software. Communication is established via CAN Remote Control Network. An UCC1 USB-CAN Converter (or e.g. a Netmax N8000 System Controller) serves as interface for linking to the PC.

Setting Up A Remote Amplifier System using a UCC1 USB-CAN Converter

When creating / installing a Remote Power Amplifier System, please make sure to keep the following information in mind:

1. Address Setting
   First, make sure to set the addresses of all Remote Power Amps with your network correctly (address selector switch on the rear panel of the power amps). CAN-networks allow addresses between 01 and 250. Set addresses and settings in the corresponding IRIS-Net project have to match.

Caution!
Assign each address only once within a specific system. Network conflicts will result otherwise.
Consequences

The default factory-settings for all Remote Power Amplifiers are:	
Parameter	**Value**
Address	00
Data transfer rate	10 kbit/s
Preset	F01 (all filters bypassed / linear, level 0dB, muting off)

2. Connecting An UCC1 USB-CAN Converter To The PC
   Connect the UCC1 USB-CAN Converter to an USB port on your PC. The STATUS LED has to light as soon as the operating system recognizes the UCC1 (IRIS-Net has not yet been started). As soon as the IRIS-Net Software has been started, the STATUS LED should blink indicating that the communication between the IRIS-Net application and the UCC1 has been established.

Caution!
Make sure to install all necessary drivers before using the UCC1. Advice on how to install the drivers is provided in the Readme-File, chapter "Installation".
Consequences

3. Network Connection Establish remote network connections (CAN-Bus) between UCC1 and all Remote Power Amps. Make sure to keep the CAN-Bus cable specifications in mind.
4. Initial Operation Make sure that no signal is present at the inputs of the remote power amps when switching their power on for the first time. Otherwise, since the power amps are set to full-range operation (F01), high output levels can result in severe damage to the connected speaker systems. To configure the Remote Power Amplifier System according to your requirements, start the IRIS-Net Software and open the corresponding project-file. Please also bear in mind the following chapters, which lay out the procedures when creating and editing IRIS-Net projects.

Setting Up A Remote Amplifier System using a NetMax N8000 System Controller
When creating / installing a Remote Power Amplifier System, please make sure to keep the following information in mind:

1. Address Setting First, make sure to set the addresses of all Remote Power Amps with your network correctly (address selector switch on the rear panel of the power amps). CAN-networks allow addresses between 01 and 250. Set addresses and settings in the corresponding IRIS-Net project have to match.

Caution!
Assign each address only once within a specific system. Network conflicts will result otherwise.
Consequences

The default factory settings for all remote power amplifiers are:	
Parameter	**Value**
Address	00
Data transfer rate	10 kbit/s
Preset	F01 (all filters bypassed / linear, level 0dB, muting off)

2. Connecting a N8000 System Controller to the PC Connect the N8000 System Controller to an Ethernet port on your PC. Please see the N8000 owner‘s manual for details.
3. Network Connection Establish remote network connections (CAN-Bus) between the N8000 and all Remote Power Amps. Make sure to keep the CAN-Bus cable specifications in mind.
4. Initial Operation Make sure that no signal is present at the inputs of the remote power amps when switching their power on for the first time. Otherwise, since the power amps are set to full-range operation (F01), high output levels can result in severe damage to the connected speaker systems. To configure the Remote Power Amplifier System according to your requirements, start the IRIS-Net Software and open the corresponding project-file. Please also bear in mind the following chapters, which lay out the procedures when creating and editing IRIS-Net projects.

Set CAN baud rate

1. Changing Baud rate of P-Series Remote Amplifiers for changing the Baud Rate to 10 kBit/s, power-off the power amp using its POWER-switch on the front panel. Disconnect the power amp from the CAN network, set its address to 00 and power-on the power amp again. Power-off the power amp and readjust the address-switches to their correct settings. Now you can reconnect the CAN network and use the power amp. This operation will reset the baud rate as well as all DSP settings to the factory-shipped defaults (F01)! Other baud rates can be selected by setting the address of the amplifier during the same procedure according to following table.

Address	**Data transfer rate**
00	10 kbit/s
Address	**Data transfer rate**
FE	62,5 kbit/s
FD	125 kbit/s
FC	250 kbit/s
FB	500 kbit/s

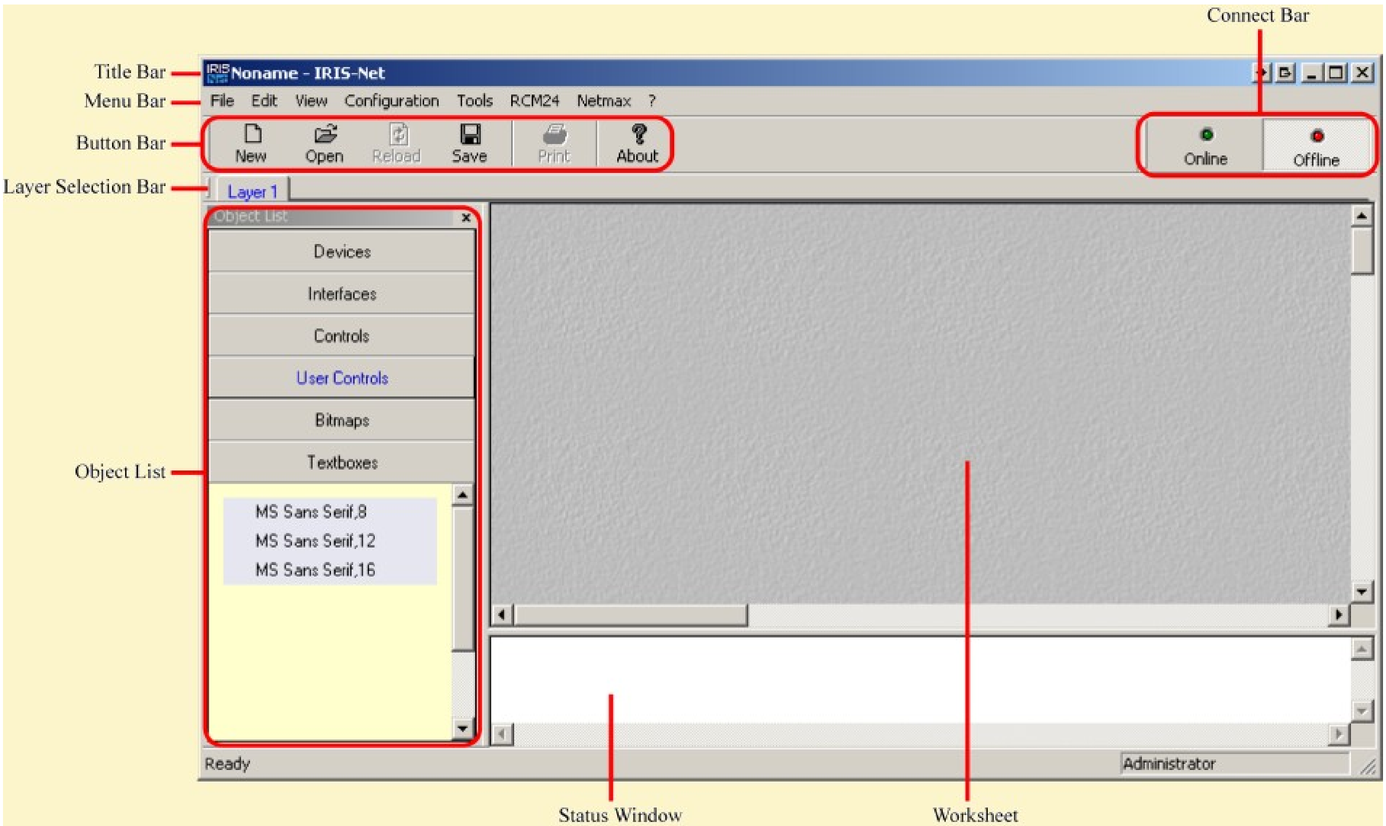
Caution!
Make sure that all Remote Power Amps within one CAN network are always set to an identical baud rate. Otherwise, if power amps within a particular network are set to different baud rates, network communications is not possible!
Consequences

2. Disable possibility to change baud rate The option Baud rate freeze allows the protection of the amplifier from a (unintended) baud rate change. For activating this option select "Modify Properties" from the amplifiers context menu in IRIS-Net. Change the value of property Baud rate freeze from 0 to 1. If Baud rate freeze is activated, the baud rate can be changed neither via Address-Switch nor via IRIS-Net. For changing baud rate you have to deactivate Baud rate freeze.

Caution!
The option Baud rate freeze must be used carefully. Activate Baud rate freeze of a amplifier only if all amplifiers at a CAN network have the same baud rate and the baud rate will not be changed in the future.
Consequences

Reference

Main Window



Element	Description
Title Bar	The title bar shows the name of the currently opened project file and IRIS-Net. "Noname" is displayed when opening a new project file or when the current project has not been saved yet. The Minimize, Maximize/Restore Size and Close/Quit IRIS-Net buttons are located on the right.
Menu Bar	Commands that can be carried out in IRIS-Net are grouped in categories. These categories are shown in the menu bar. Clicking on one of these categories displays the corresponding command list.
Tool Bar	Frequently used commands are arranged as buttons in the tool bar. The button of a command that is not available is grayed-out. Clicking the button has no effect.
Layer Selection Bar	If a project file includes various layers, switching between these layers is possible by selecting the corresponding tab in the Layer Selection Bar. Left-clicking on a previously selected tab allows renaming this tab. Right-clicking on a tab opens a context menu which allows changing the order of the layers within the project.
Object List	The Object List contains all categories of objects that can be arranged in the worksheet.
Connect Bar	The Connect Bar provides access to the on-line dialog.

Status Window	The Status Window displays IRIS-Net system status messages. Additionally, the name of the currently logged-in user is shown underneath the Status Window.
Worksheet	In the Worksheet, the user can arrange control and display panels of a project by dragging objects out of the Object List and dropping them into the Worksheet.

Menus, Commands and Symbol bar

A context menu offers a limited set of choices that are available in the current state, or context, of the object, device or application.

1. Position the cursor over the desired object.
2. A click with the right mouse button opens the context menu at the spot where you have clicked.

Menu “File”

Element	Shortcut	Description
New	Ctrl+N	Creates a new project file.
Open...	Ctrl+O	Loads an existing project.
Reload...	Ctrl+R	Reloads the current project.
Save	Ctrl+S	Saves a project.
Save As...		Saves an existing project under another name.
Print Setup...		Change the printer and printing options.
Last used project files		The last four projects used are listed here.
Exit		Shuts the program down.

Menu “Edit”

Element	Shortcut	Description
Cut	Ctrl+X	Removes the selected item(s) and places them on the clipboard.
Copy	Ctrl+C	Places a copy of the selected item(s) on the clipboard.
Paste	Ctrl+V	Insert item(s) from the clipboard.
Delete	Del	Deletes the selected item(s).
Settings...		Opens the Settings dialog.
Device Options...		Opens the IRIS-Net Device Options dialog.
Network Settings...		Opens a dialog to select the Ethernet interface (e.g. network card of the PC) to use for OMNEO or Ethernet devices.

Menu “View”

Element	Description
Toolbars	Shows the toolbar below the menu bar.

Status Bar	Shows the status bar at the bottom of the IRIS-Net workspace window.
Object List	Shows the object list at the left of the IRIS-Net workspace window.

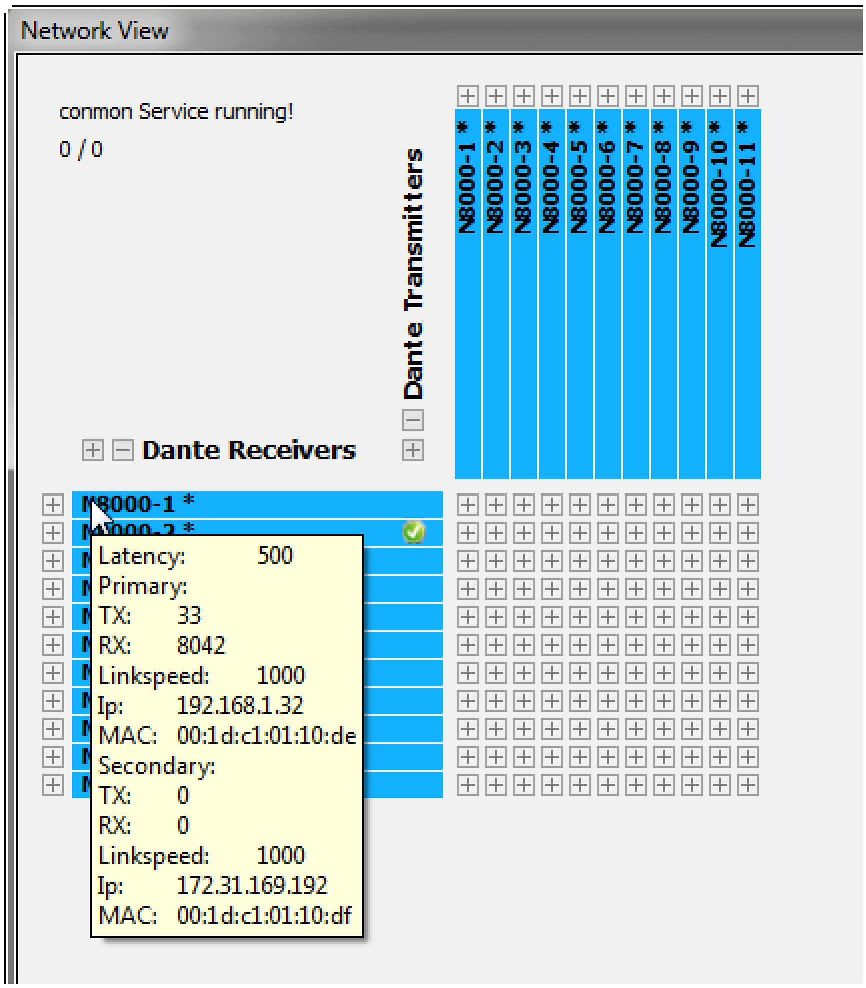
Menu “Configuration”

Element	Description
Add Device...	A click on Add Device opens the IRIS-Net Device List allowing you to drag new devices into the worksheet area.
Add Interface...	A click on Add Interface opens the IRIS-Net Interface List allowing you to drag new interfaces into the worksheet area.
Add Control...	A click on Add Control opens the IRIS-Net Controls List and lets you create new controls.
Add User Controls...	A click on Add User Controls opens the User Controls List and allows adding predefined or user-defined control panels to the project.
Add Bitmap...	Select Add Bitmap to add any of the supplied or self-designed bitmaps to your IRIS-Net project.
Add Textbox...	Select Add Textbox to open the dialog box for text entries.
Modify Properties	Modify Properties opens the IRIS-Net Properties Dialog where you can make or change basic settings of a project.
Add Layer	A click on Add Layer lets you add a new page (layer) to the IRIS-Net project. A new register (Layer x) is created in the IRIS-Net worksheet. Assign a meaningful name to the new register, e.g. "Control Page". You can lay out the new page and add controls according to your preferences. IRIS-Net allows the use of up to 32 layers.
Delete Layer	A click on Delete Layer deletes the currently active page (layer) from the IRIS-Net project. **CAUTION:** **Delete Layer erases the entire contents of the currently active page. However, as long as the project is not saved, all data is still present in the project file. If you inadvertently have deleted a page, use Reload to revert to the last saved project state. Once you have saved your project after deleting a page, all information/data of that specific page is gone.**
Passwords...	A click on Passwords opens the IRIS-Net Password Dialog which lets you define access rights to your project.
Logout	Click on Logout to log out of the actual IRIS-Net project. Now you can log in again using a different password and different access rights if these have been configured in the project.

Menu “Tools”

Element	Description
SPL Calculator	Opens the window SPL Calculator.
Limiter Threshold Calculator	Opens the window Limiter Threshold Calculator
Event Scheduler	Opens the window Event Scheduler.
Macro Editor	Opens the window Macro Editor. There are the tabs Macros, Triggers and Conditions.
Scene Editor	Opens the window Scene Editor.
Event Log	Opens the IRIS-Net Event Log.
Device Scan	Opens the window Device Scan.

PA Event Log	Opens the window PA Event Log
OCA Scan	Opens the window OCA Scan.
Dante Configuration	Opens the window Network View. In this window you can edit the routing in your Dante network.



Menu “RCM-24”

Element	Description
Configuration via CAN Hardware	Opens the window Configuration
Control Functions	Opens the window Control Functions.
System Check	Opens the window RCM-24 System Check.
Overview	Opens the window RCM-24 Overview.

Menu “RCM-26”

Element	Description
Configuration via CAN Hardware	Opens the window Configuration

System Check	Opens the window RCM-26 System Check
Overview	Opens the window RCM-26 Overview.

Menu “RCM-810”

Element	Description
Overview	Opens the window RCM-810 Overview.

Menu “RCM-28”

Element	Description
System Check	Opens the window RCM-28 System Check
Overview	Opens the window RCM-28 Overview.

Menu “Matrix”

Element	Description
Configuration via USB	Opens the window NetMax Configuration via USB.
Real Time Clock	Opens the window Set N8000 Real Time Clock for editing real time clock of all NetMax devices in the project
Superblocks	Opens the window Matrix Superblocks for deleting Superblocks.

Menu “?”

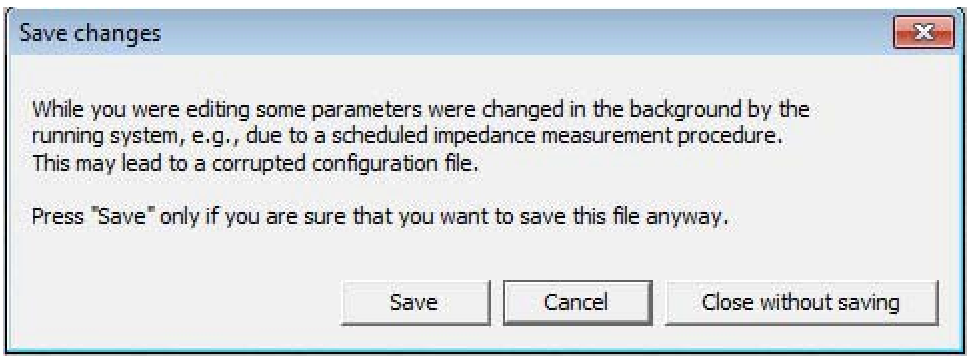
Element	Description
Project Info	Opens the window Project Info
Help Topics	Opens IRIS-Net help (this document)
Quick Start Guide...	Opens the IRIS-Net Quick Start Guide.
Search for Updates ...	Opens the Software Update dialog.
About IRIS....	Shows the current application software version.

Symbol Bar

Element	Description
New	Creates a new project file.
Open	Loads an existing project.
Reload	Reloads the current project.

Save	Saves a project.
About	Lists the current application software version.
Online/Offline	If current status is offline, Online opens the window Going online.

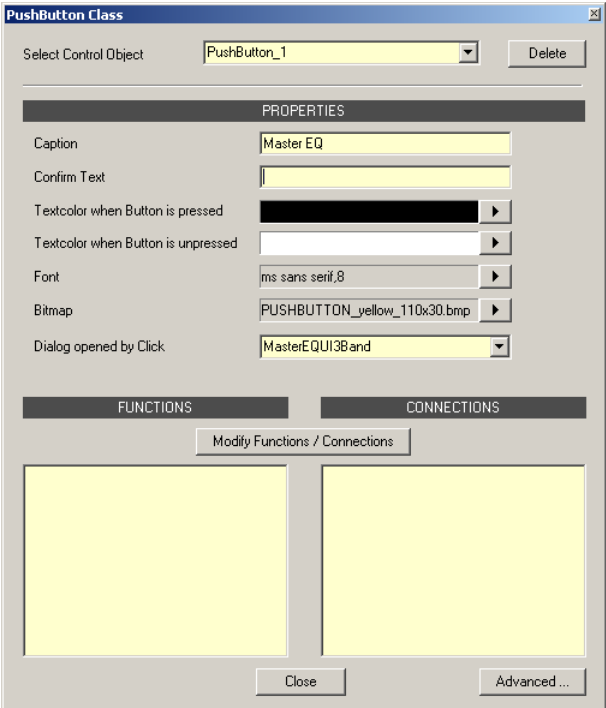
Please note following hint concerning the Going online dialog:
When being online with a DPM 8016, the IRIS-Net Configuration reads out system parameters which may differ from the default settings (e.g. running impedance measurement). If these changes are coming from other devices except the local PC, and the configuration file shall be saved after being online with the system, the user is asked if he wants to keep the original version or take the current version including the modified parameters.



Controls

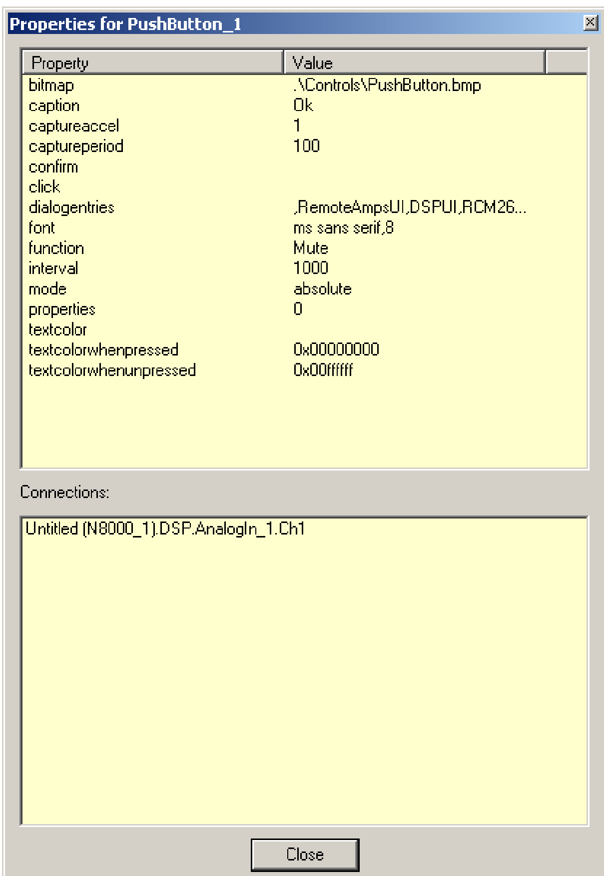
PushButton

The IRIS-Net Control of the type "PushButton" is the equivalent of a button. The button has two states: pressed and unpressed. Left-clicking on the "PushButton" Control activates the "pressed" state. The "unpressed" state is restored immediately after releasing the mouse button. The button's appearance in both states can be freely configured. Activating a safety dialog when pressing the button is additionally possible. One or more parameters that are activated in the "pressed" state can be assigned to a PushButton.



Element	Description
Select Control Object	Switching between Controls for editing is possible if a variety of Controls of the type "PushButton" are used in a project.
Delete	The currently selected PushButton is deleted in the IRIS-Net project.
Caption	Text label of the PushButton. The entered text is always displayed centered.
Confirm Text	If it is intended that a safety dialog appears when left-clicking on the button, this field lets the user enter an explanatory text to be displayed in that safety dialog. If no text is being entered, the safety dialog does not appear.
Textcolor when Button is pressed	Color of the button label when in the "pressed" state. The button opens the Color dialog which allows selecting between predefined and user-definable colors.
Textcolor when Button is unpressed	FColor of the button label when in the "unpressed" state. The button opens the Color dialog which allows selecting between predefined and user-definable colors.
Font	Displays the currently selected typeface for button labeling. The button opens the "Typeface" dialog which allows selecting font type, font style/weight, and font size.
Bitmap	Displays the filename of the bitmap graphic file that is currently used to represent the pushbutton. The button opens the "Bit- maps" dialog which allows selecting the desired bitmap file to represent the pushbutton.
Dialog opened by Click	Allows selecting a dialog that opens when pressing the PushButton.
FUNCTIONS	Lists the functions (WHAT-part of the parameter) that are active when the PushButton is in the "pressed" state. Removing items is possible by opening the contextual menu of the function that is to be deleted and selecting the entry "Delete Entry".

CONNECTIONS	Lists the devices/objects (WHERE-part of the parameter) for which the selected functions will be active when the PushButton is in the "pressed" state. Removing items is possible by opening the contextual menu of the Connection that is to be deleted and selecting the entry "Delete Entry".
Modify Functions / Connections	Opens the "Modify Functions & Connections" dialog that lets the user select the parameter(s) of the PushButton to be modified.
Close	Closes/cancels the "PushButton Class" dialog.
Advanced ...	Opens the "Properties for PushButton" dialog.



Property	Description
bitmap	Path and filename of bitmap used for button. Path must be entered relative to \IRIS-Net.
caption	Caption of button
Capture accel	Only used if mode is "increment" or "decrement". Every capture period ms the new value is calculated by: new_value = old_value * capturea- ccel
Capture period	Every capture period ms the state of the button is polled.
confirm	Text of confirm dialog.

click	Action that should be executed when the PushButton is clicked. See following table Click for details.
Dialog entries	List of dialogs included in Drop Down Dialog opened by Click.
font	Font type and font size, separated by comma.
function	Active function if button is in pressed state. Multiple functions are separated by comma.
interval	The parameter connected with the PushButton is polled every interval ms. Enter "0" if the parameter should not be polled.
mode	absolute for 0/1 at state unpressed/pressed increment/decrement for increase/decrease of parameter during state pressed
properties	If mode = absolute: State of push button (0 = unpressed, 1 = pressed) If mode = increment or decrement: Lower border of influenced parameter.
Text color	deprecated
Text color when pressed	Color of caption in state pressed, given in hexadecimal format:: 0x00BBGGRR with BB = blue, GG = green, RR = red
Text color when unpressed	Color of caption in state unpressed, given in hexadecimal format:: 0x00BBGGRR with BB = blue, GG = green, RR = red

Property click.

Type	**Format**	**Example**	**Comment**
Execute a file	doc*open=<path and file name>	doc*open=c:\alarm.mp3	Every file type can be executed, Microsoft Windows uses the default application (depending on file extension) for executing the file (e.g. *.pdf --> Adobe Reader, *.mp3 --> Winamp). Absolute or relative path description is possible.
Set parameter value	<Device>*<Keyword>=<Value>	N8000_1*DSP.AnalogIn_1.Ch3.Mute=1	Setting illegal parameters should be avoided.
Change layer	changelayer=<number OfLayer>	changelayer=3	Moving layers (via context menu "Move to left/right") does not change its number, find number via property "layeractive" of worksheet properties.
Execute a script	script=<path and file name>	script=.\Scripts \SeparateRooms.dss	Absolute or relative path description is possible.
Execute a scene	script=<path and file name>	script=.\Scenes \SeparateRooms.scn	Absolute or relative path description is possible.
Open a template	template=<templateName> <"Window Title"> %c	template=N8000_PEQ_5 Band_Mono "PEQ 5 Band" %c	See directory \IRIS-Net\Templates for available tem- plates. "Window Title" is displayed in the title bar of the template window..

Going online/offline	*online=<0/1>	*online=1	use 1 for going online, use 0 for going offline
Change user of IRIS-Net	Change current user=<UserName>	Change current user=NewUser1	Entering the password of the user is not required when "logging in" via PushButton. This option should be used for changing set of accessible layers only.
Fullscreen/window mode	fullscreen=<yes/no>	fullscreen=yes	Use "yes" for fullscreen mode, use "no" for window mode.
Execute channel A impedance test of RCM- 26 amplifier	chasweep=0/<number of measurements>: 1 <start freq> Hz, <stop freq> Hz, 0 s, <level> dB, solo, post	chasweep=0/150: 1 10000 Hz, 20000 Hz, 0 s, -10 dB, solo, post	The impedance test is executed at channel A of the RCM-26 remote amplifiers connected to the button.
Execute channel B impedance test of RCM- 26 amplifier	chbsweep=0/<number of measurements>: 1 <start freq> Hz, <stop freq> Hz, 0 s, <level> dB, solo, post	chbsweep=0/150: 1 10000 Hz, 20000 Hz, 0 s, -10 dB, solo, post	The impedance test is executed at channel B of the RCM-26 remote amplifiers connected to the button.
Execute impedance test (RCM-24)	impedancecurve=0/0: 0 <start freq> Hz, <stop freq> Hz, 200 ms, <level A> dB, <level B> dB, solo, post	impedancecurve=0/0: 0 20 Hz, 20000 Hz, 200 ms, -10 dB, -10 dB, solo, post	The impedance test is executed at the RCM-24 remote amplifiers connected to the button.
Loads a preset in the device(s) (RCM-24 amplifier, RCM-26 amplifier, Dx46/DSP 600 or N8000/P 64) connected to the PushButton.	*LoadPreset=<preset number>	N8000 or P 64: *LoadPreset=01 Remote Amplifier or Dx46/ DSP 600: *LoadPreset=U01	Possible connections are the remote amplifier, the Dx46 or DSP 600, or N8000_x.DSP or P64_x.DSP.
Saves a preset (or scene) in the device(s) (RCM-24 amplifier, RCM-26 amplifier, Dx46/DSP 600 or N8000/P 64) connected to the PushButton.	*SavePreset=<preset number>	N8000 or P 64: *SavePreset=01 Remote Amplifier or Dx46/ DSP 600: *SavePreset=U01	Possible connections are the remote amplifier, the Dx46 or DSP 600, or N8000_x.DSP or P64_x.DSP.

Templates

A control panel for a single or several device(s) can be accessed via a Group or a Push Button. Call up the contextual menu of a Group or a Push Button and select the desired control panel under the menu item "Dialog opened by Click". General information on how to use Groups is provided in the chapter "Working with Groups". Available control panels are listed in the following table:

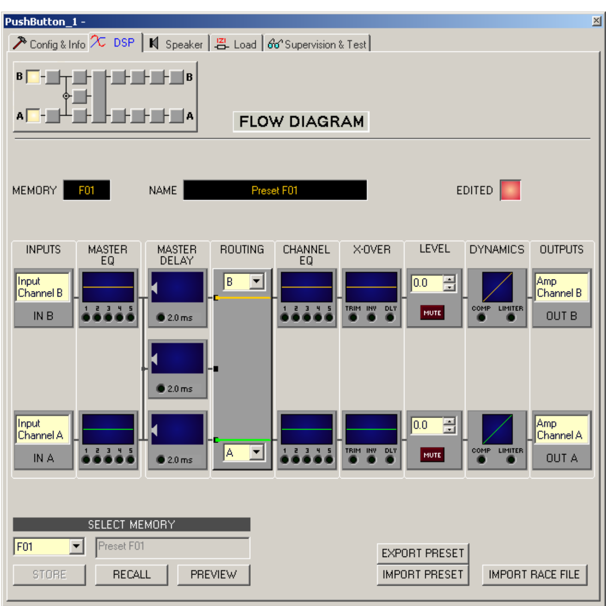
Template	Description
ArrayPEQ_5band_UI	
DSP600DSPUI	
DSP600UI	
DSP600Userpanel	
DSPUI	
DX38UI	
DX46DSPUI	
DX46UI	
DX46Userpanel	
InputGEQ_31band_UI	
InputPEQ_10band_UI	
Master EQ UI x Band	x = 3, 4, 5, 6
N8000_Delay_xms	x = 10, 100, 500, 2000
N8000_GEQ_xWay_Mono	x = 10, 15, 31
N8000_LSC_xWay_Mono	x = 1, 2, 3, 4, 5
N8000_PEQ_xBand(_Mono)	x = 3, 5, 7 ,12
N8000_ToneControl_Mono	
N8000_ToneGenerator	
OutputPEQ_6band_UI	
RCM26DSPUI	
RCM26RemoteAmpsUI	
RCM28DSPUI	
RCM28RemoteAmpsUI	
Remote Amps UI	
System Crossover_xWay	x = 4,5,6,7,8

The description of the controls or indicators included in the templates is given in the section of the corresponding device. A selection of templates is described on the following pages.

Remote Amps UI

The "Remote Amps UI" control panel is the equivalent of the Setup & Control Window of an RCM-24 Remote Amplifier. The control panel is accessed via a Push Button or a Group and is used for the synchronized control of various amplifiers. The RCM-24 Remote Amplifiers to be controlled have to be linked to the Push Button or the Group via the menu item "Administrate Connection" of the corresponding contextual menu.
For more detailed information please refer to the corresponding chapters of the Remote Amplifier.

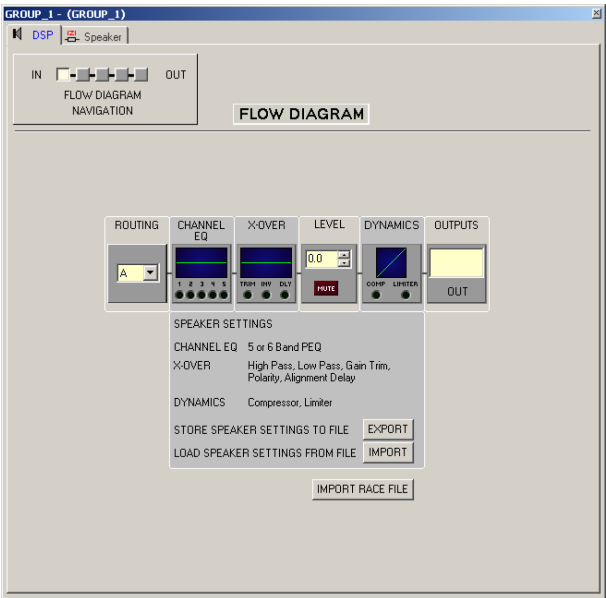
Tab	Description
Config & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Filter, Delay, X-Over, and Dynamics) of the amplifier.
Speaker	This page allows loading and displaying speaker data.
Load	This page provides access to several settings for impedance/load monitoring and impedance testing.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.

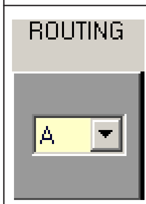
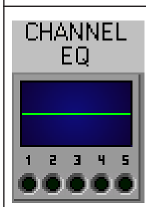
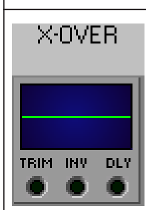


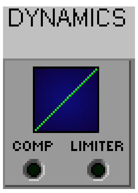
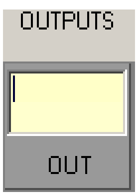
Single Channel Group Dialog (DSPUI)

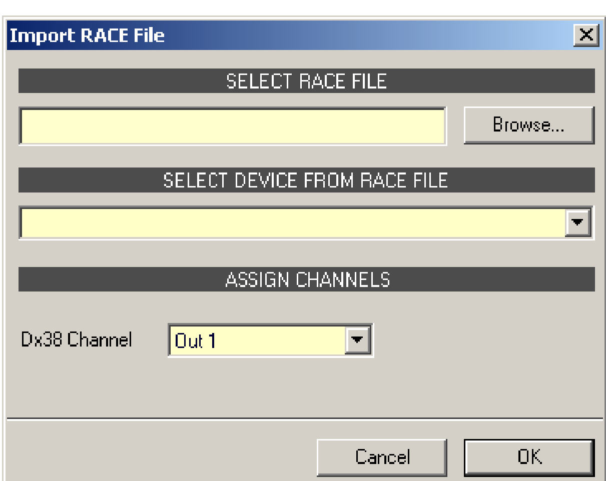
The Single Channel Group Dialog (for RCM-24 remote amplifiers) is used for linearization and secure operation of a single loudspeaker. An equalizer, a crossover and a compressor/limiter are presented in separate window. This allows comfortably adjusting all the specific settings that are necessary for the operation of a single loudspeaker chassis (or individual way of a loudspeaker system) in a separate window. All settings/parameters can be saved in configuration files (Speaker Settings). Pre-defined configuration files (Speaker Settings) are available for loudspeakers from Electro-Voice and DYNACORD . The Single Channel Group Dialog can be accessed via a Group or a Push Button. In the properties of the respective element you have to select the setting "DSPUI" in the dropdown field "Dialog opened by Double Click". The channels of the RCM-24 Remote Amplifiers to be controlled have to be linked to the Push Button or

the Group via the menu item "Administrate Connection" of the corresponding contextual menu. The two tabs "DSP" and "Speaker" can be found in the main window of the Single Channel Group Dialog. The DSP tab provides access to the flow diagram to make all necessary settings.



Element	Description
	The flow diagram selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.
	Select A or B if the setting should be valid for the corresponding channel of the connected amplifier. Select A and B if the setting should be valid for both amplifier channels.
	The Channel EQ block displays the 5 Channel EQs of the corresponding output channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse but- ton onto this block opens the X-OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X-Over block within the same project.

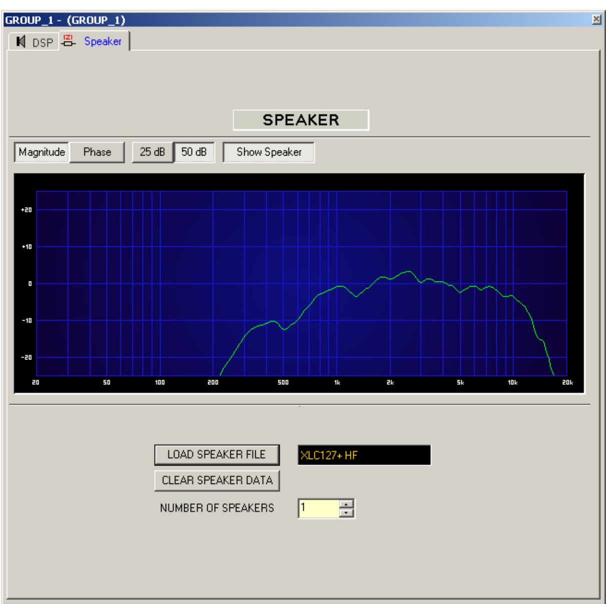
	The numerical field is identical to the numerical field below the level controls in the Amplifier Control Panel. So the field indicates the actually set attenuation, by which the internally specified amplification is attenuated, in dB. The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	This block provides graphical display of the dynamics functions of the corresponding output. The two LEDs indicate whether compressor or limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Dynamics block to any other Dynamics block within the same project.
	The text field allows specifying a name for the corresponding output channel.
EXPORT SPEAKERSETTING	After clicking onto EXPORT SPEAKERSETTING a Save File dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store the speaker setting together with the corresponding file name.
IMPORT SPEAKERSETTING	After clicking onto IMPORT SPEAKERSETTING an Open File dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file.
IMPORT RACE FILE	IRIS-Net allows importing loudspeaker presets that have been created in Electro-Voice RACE by clicking onto the button IMPORT RACE FILE to open the following dialog box.



First, you have to select the desired RACE file by use of the Browse... button. Because a RACE file can hold the data of up to 31 EV Dx38, you need to continue by selecting the desired device from the RACE file within the dialog SELECT DEVICE FROM RACE FILE. At the end you have to specify which of the four Dx38 output channels should be assigned to the corresponding amplifier channels. Clicking onto OK button completes the process.

Caution!
The loaded RACE file becomes instantly audible when in on-line mode. Be sure to select the desired file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the amplifier channels and display the acoustic results of this virtual combination. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of all common Electro-Voice and DYNACORD loudspeaker systems. Some examples are provided in the IRIS-Net directory Speaker Files.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the amps. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters.
Clicking on the Speaker tab in the DSPUI window opens the Speaker page.

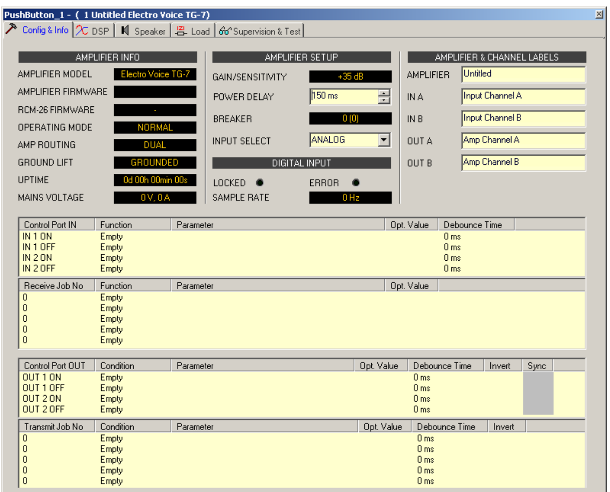


RCM26RemoteAmpsUI
The "RCM26RemoteAmpsUI" control panel is the equivalent of the Setup & Control Window of the Remote Amplifiers with installed RCM-26 module.
The control panel is accessed via a Push Button or a Group and is used for the synchronized control of various amplifiers. The Remote Amplifiers to be controlled have to be linked to the Push Button or the Group via the menu item "Administrate Connection" of the corresponding contextual menu.

Tab	**Description**
Config & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.

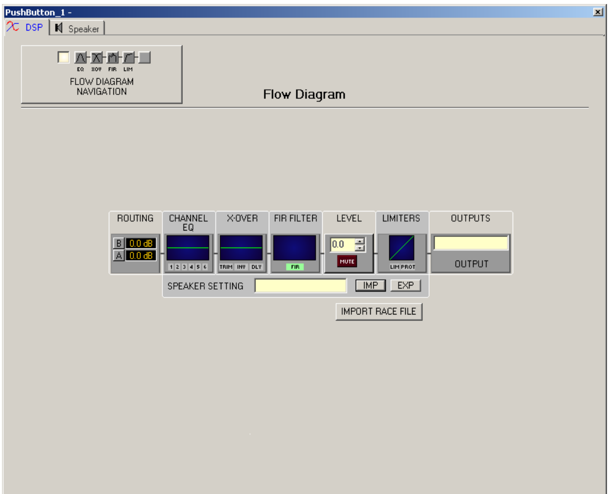
DSP	The DSP page provides an overview plus access to all DSP functions (Filter, Delay, X-Over, Limiters) of the amplifier.
Speaker	This page allows loading and displaying speaker data.
Load	This page provides access to several settings for impedance/load monitoring and impedance testing.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.

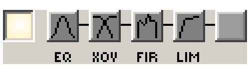
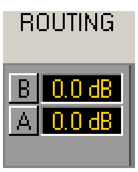
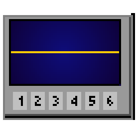
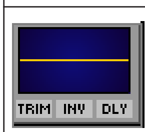
For more detailed information please refer to the corresponding chapters of the Remote Amplifier.

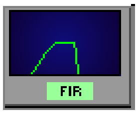
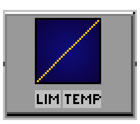
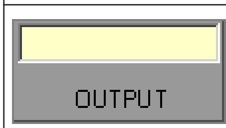


RCM26DSPUI

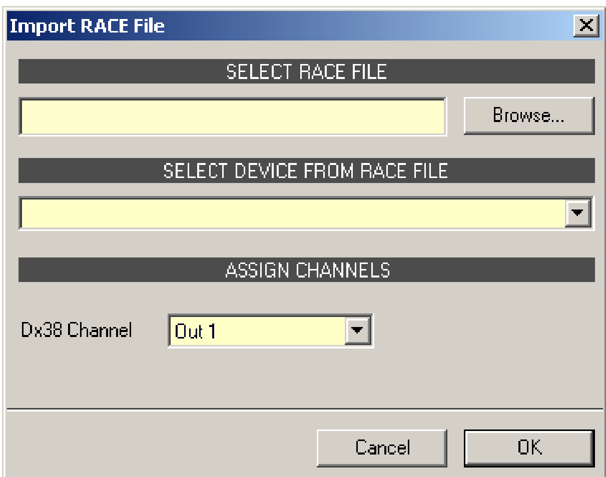
The RCM26DSPUI Dialog is used for linearization and secure operation of a single loudspeaker. An FIR-filter, equalizer, a crossover and limiters are presented in separate window. This allows comfortably adjusting all the specific settings that are necessary for the operation of a single loudspeaker chassis (or individual way of a loudspeaker system) in a separate window. All settings/parameters can be saved in configuration files (Speaker Settings). Pre-defined configuration files (Speaker Settings) are available for loudspeakers from Electro-Voice and DYNACORD.
The RCM26DSPUI Dialog can be accessed via a Group or a PushButton. In the properties of the respective element you have to select the setting "RCM26DSPUI" in the dropdown field "Dialog opened by Double Click". The two tabs "DSP" and "Speaker" can be found in the main window of the RCM26DSPUI Dialog. The DSP tab provides access to the flow diagram to make all necessary settings.



Element	Description
	The flow diagram selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.
	Here you can assign the output channel routing. The A and B buttons allow selecting the input signal for the corresponding output channel. Clicking with the right mouse button onto the dB display opens a fader.
	Channel EQ Block: The Channel EQ block displays the 6 Channel EQs of the corresponding output channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Crossover Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse button onto this block opens the X- OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X-Over block within the same project.

	FIR Filter Block: This block represents the FIR Filter within the corresponding output channel. The graph shows the frequency response that results from the set FIR parameters. The LED indicate if the FIR Filter is being used. A single click with the left mouse button onto this block opens the FIR page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding FIR Filter block to any other FIR Filter block within the same project.
	Level Block: The numerical field is identical to the numerical field below the level controls in the Amplifier Control Panel. So the field indicates the actually set attenuation, by which the internally specified amplification is attenuated, in dB. The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Limiters Block: This block provides graphical display of the limiters functions of the corresponding output. The two LEDs indicate whether peak limiter or thermal limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Limiters block to any other Limiters block within the same project.
	Output Block: The text field allows specifying a name for the corresponding output channel. A click with the right mouse button onto OUT A or OUT B opens the Copy & Paste menu, which allows copying all parameters of the corresponding output channel (Routing, Channel EQ, X-Over, FIR Filter, Dynamics) to any other output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied. Allows import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for DSP blocks Channel EQ, Crossover, FIR Filter and Dynamics & Protect. The text field allows editing the description of the Speaker Setting file.

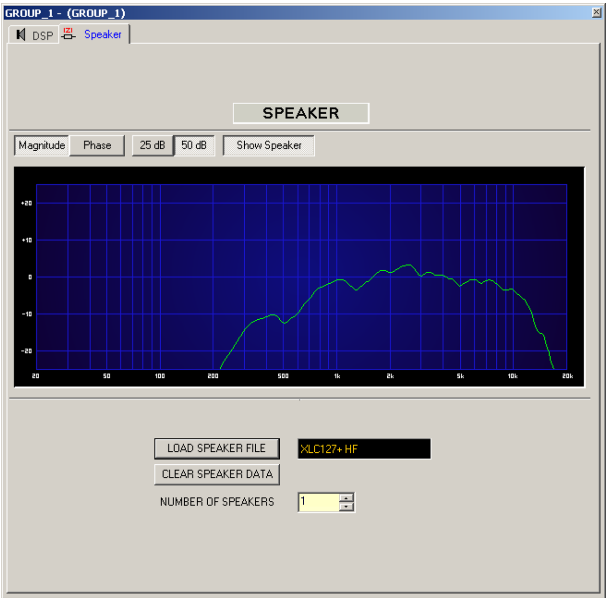
	Allows import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for DSP blocks Channel EQ, Crossover, FIR Filter and Dynamics & Protect. The text field allows editing the description of the Speaker Setting file.
	IRIS-Net allows importing loudspeaker presets that have been created in Electro-Voice RACE by clicking onto the button IMPORT RACE FILE to open the following dialog box.



First, you have to select the desired RACE file by use of the Browse... button. Because a RACE file can hold the data of up to 31 EV Dx38, you need to continue by selecting the desired device from the RACE file within the dialog SELECT DEVICE FROM RACE FILE. At the end you have to specify which of the four Dx38 output channels should be assigned to the corresponding amplifier channels. Clicking onto OK button completes the process.

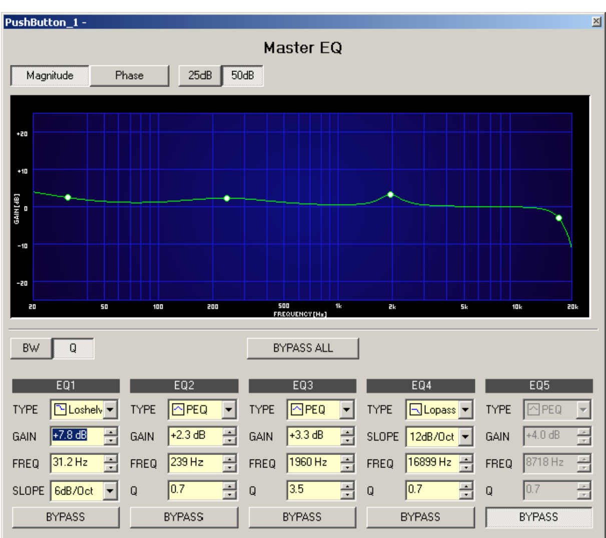
Caution!
The loaded RACE file becomes instantly audible when in on-line mode. Be sure to select the desired file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the amplifier channels and display the acoustic results of this virtual combination. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of all common Electro-Voice and DYNACORD loudspeaker systems. Some examples are provided in the IRIS-Net directory Speaker Files.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the amps. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters.



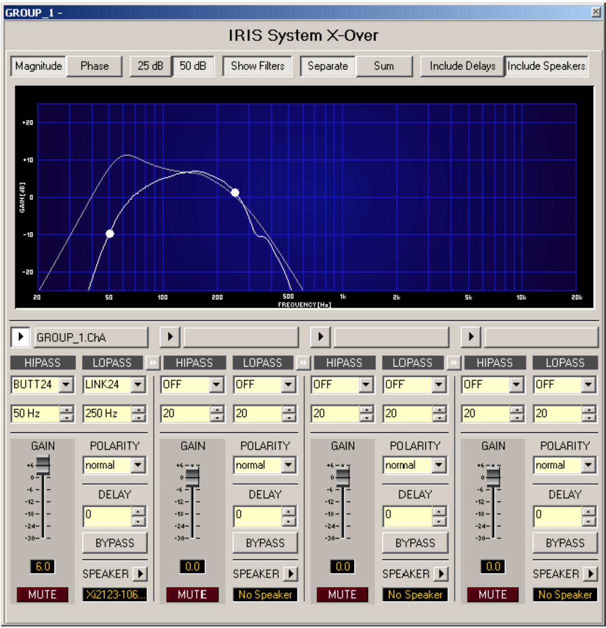
Master EQUI x Band

The "Master EQ" control panel is available in three different versions - offering three, four, or five band EQs. Any individual Remote Amplifier's equalizer band that exists in the IRIS-Net project may be assigned to each of the Master EQ bands. The control panel is accessed via a Push Button or a Group. The desired devices with equalizer bands to be controlled assigned to, have to be linked to the Push Button or the Group via the menu item "Administrate Connection" of the corresponding contextual menu.



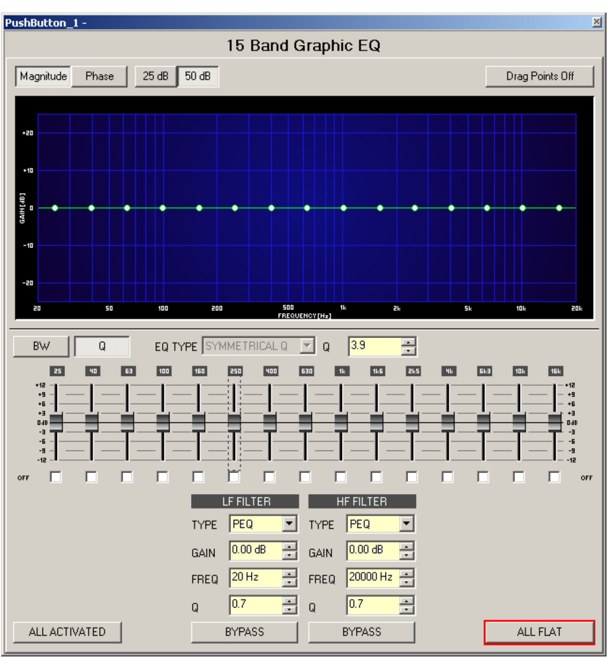
System Crossover_x Way

The "IRIS System X-Over" control panel features several independent frequency crossover channels. Any individual crossover channel that exists in the IRIS-Net project may be assigned to each of the System X-Over bands, which allows controlling the crossovers of a specific amplifier as well as those of various units from a single window. The control panel is accessed via a Push Button or a Group. The devices with crossovers to be controlled assigned to, have to be linked to the parent object (the accessing object) via the menu item "Administrate Connection" of the corresponding contextual menu.



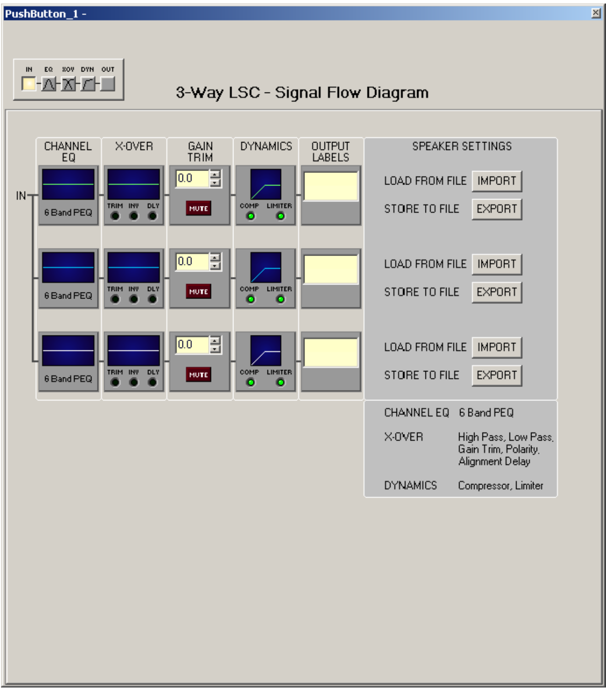
N8000_GEQ_xWay

The "N8000_GEQ_xWay" control panel is available in three different versions - offering 10, 15, or 31 band GEQs. It is the equivalent of the DSP Block Window "GEQ" of a N8000. The control panel is accessed via a Push Button or a Group and is used for the synchronized control of several N8000. The GEQs to be controlled have to be linked to the Push Button or the Group via the menu item "Administrate Connection" of the corresponding contextual menu.



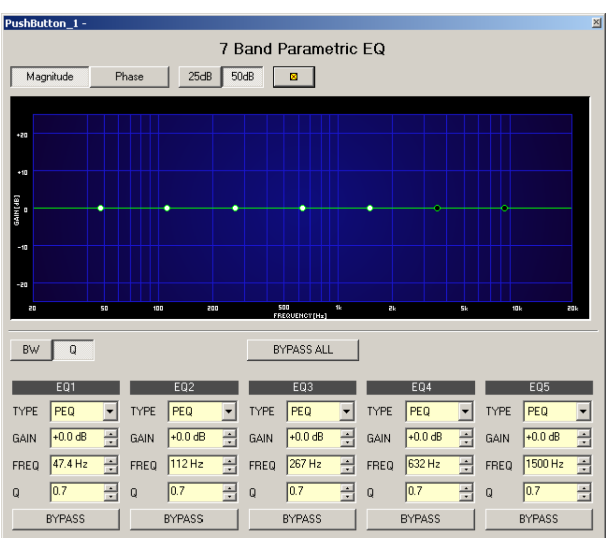
N8000_LSC_xWay_Mono

The "N8000_LSC_xWay" control panel is available in five different versions - offering one, two, three, four, or five channels. It is the equivalent of the DSP Block Window "Monaural Loudspeaker Controller" of a N8000. The Loudspeaker Controllers to be controlled have to be linked to the Push Button or the Group via the menu item "Administrate Connection" of the corresponding contextual menu.



N8000_PEQ_xBand (_Mono)

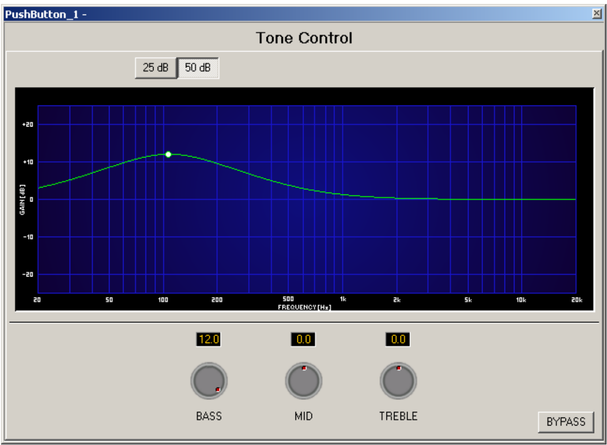
The "N8000_PEQ_xBand_Mono" control panel is available in five different versions - offering three, five, seven, or twelve band PEQs. It is the equivalent of the DSP Block Window "Mono PEQ" of a N8000.



Element	Description
	Press this button to update the settings of all connected PEQs.

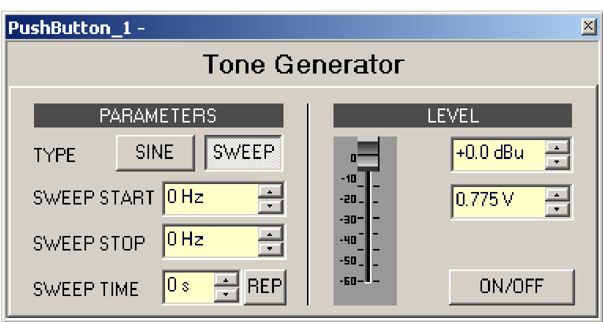
N8000_ToneControl_Mono

The "N8000_ToneControl_Mono" control panel is the equivalent of the DSP Block Window "Tone Control Mono" of a N8000.



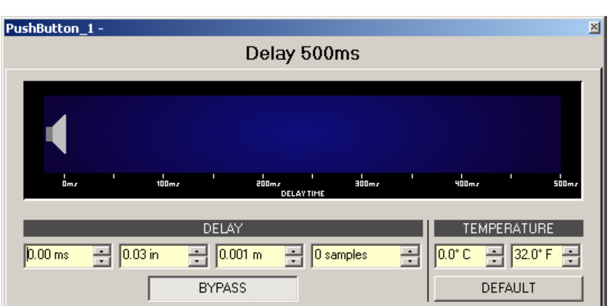
N8000_ToneGenerator_Mono

The "N8000_ToneGenerator_Mono" control panel is the equivalent of the DSP Block Window "Tone Generator Mono" of a N8000.



N8000_Delay_xms

The "N8000_Delay_xms" control panel is the equivalent of the DSP Block Window "Mono Delay 10/100/500/2000 ms" of a N8000.



REMOTE AMPLIFIER

P-Series

Introduction

The IRIS-Net software (Intelligent Remote & Integrated Supervision) runs under Windows and allows configuring, controlling and monitoring a complete PA-system from a single or from several PCs. Any operational status, e.g. power-on, temperature, level, limiting, activation of protections, deviation of the output impedance, etc., are centrally recorded and displayed, which offers the opportunity to react and interfere even before the occurrence of critical operational states. Programming automated actions that are carried out when exceeding or falling short of certain threshold values is possible as well. All parameters, e.g. power-on/off, level, mute, filters, etc. are controlled in real-time and can be stored in any power amplifier.
Monitoring the connected loudspeaker systems is performed by continuously measuring output currents and voltages of individual power amplifier channels. Each exceeding or falling short of set thresholds is instantly signaled and logged. In this way, short-circuits or line interruptions, as they might occur during normal operation, are recognized and displayed immediately. The integrated impedance test function allows checking the connected loudspeaker systems more precisely. Together with the current/voltage testing function the integrated sweep-generator is employed to measure the connected loudspeakers’ and cables’ impedance over the entire frequency range. The resulting impedance graph is displayed on the PC-screen. Comparing the measured impedance progression with a reference value is possible at all times, which allows recognizing even the slightest defects or irregularities of the loudspeaker systems.
Next to its controlling and monitoring capabilities, the IRIS-Net System provides comprehensive signal processing functions. A total of 20 parametric filters, X-over functions, delays, routing and level control as well as compressors and limiters per each channel are included. All parameters can be freely edited and stored in up to eight user presets within the remote amplifiers. Independent from network control all DSP settings (filter, delay, level) are maintained incase of failure. Additionally, the control inputs of the power amps can be used for network-independent switching to another preset (e.g. alarm settings with maximum energy for voice and text announcements).
Therefore, the IRIS-Net System fulfills even the highest safety requirements.

Remote Power Amps

Electro-Voice PRECISION SERIES REMOTE CONTROL Series of power amps employ most advanced technologies offering a truly remarkable combination of outstanding audio performance, highest reliability and operational stability. The gapless protection circuitry concept not only protects each power amp itself but also prevents the connected loudspeaker systems from being damaged. These extensive protections include Dynamic Audio Limiters, DC/HF-Protections, Back-EMF-Protection, Inrush Current Limiter, Short Circuit Protection and of course Thermal Overload Protection for the output transistors and mains transformers.
Three-speed high performance fans guarantee outstanding thermal stability at absolute low running noise. The ventilation is directed front-to-rear allowing trouble-free operation even in smaller amp-racks.
Comprehensively dimensioned power supply units with low-leakage toroidal transformers provide extensive headroom far above the stated nominal power. Mechanical construction and workmanship also comply with the highest precision manufacturing standards. The rigid sheet steel chassis resists even the most wearing touring operation.
The Electro-Voice PRECISION SERIES REMOTE CONTROL Series consists of the following remotely controlled power amps:

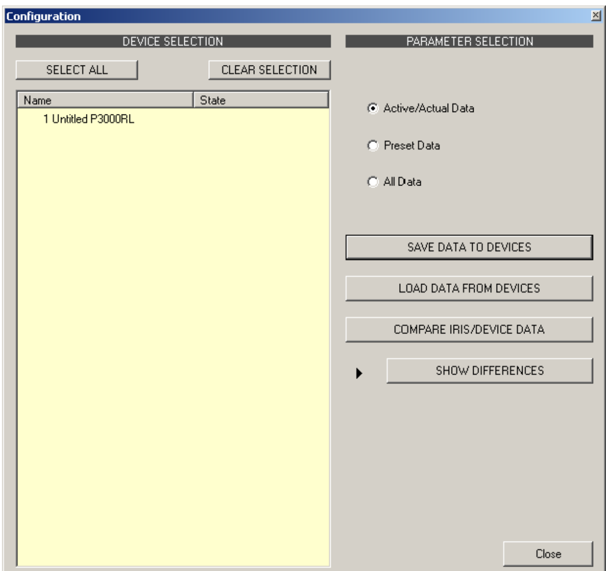
- P3000RL 2 x 1300 W / 4 Ohm
- P1200RL 2 x 600 W / 4 Ohm
- P900RL 2 x 450 W/ 4 Ohm
- P1200RT 2 x 590 W / 100 V
- P900RT 2 x 410 W/ 100 V

Each power amplifier employs an RCM-24 Remote Control module, which allows integration into a Remote Control Network consisting of up to 100 power amps. Detailed information on the individual modules is provided in the corresponding manuals.

How to...

CHECKING/EDITING THE CONFIGURATION

Each remote amplifier in a project has an RCM-24 (Remote Control Module) installed. A PCI/USB-CAN-Interface can be used to connect the CAN-Interface of an RCM-24 to the PC. Existing RCM-24 modules are configured in the Configuration Window. When using a PCI-CAN-Interface or USB-CAN-Interface, the window needs to be opened via the menu item "RCM-24 | Configuration via CAN Hardware".
Devices to be configured can be chosen in the left side of the window while the right side of the window selects the data to be viewed during the configuration procedure.



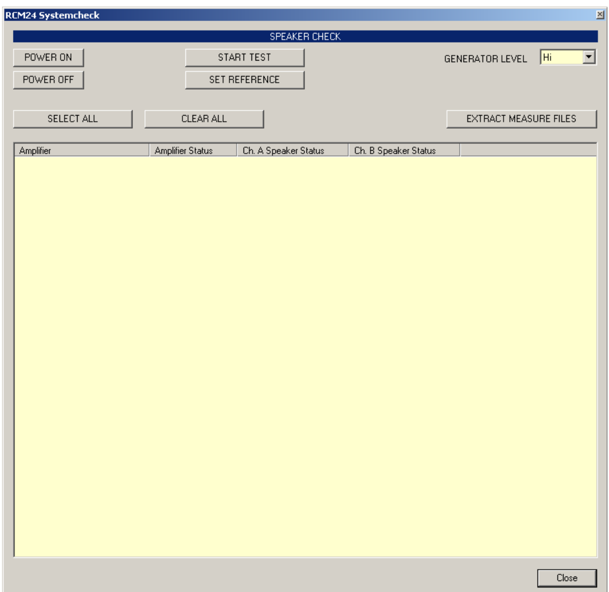
Element	Description
SELECT ALL	All devices in the list are selected for configuration.
CLEAR SELECTION	Resets the current selection of devices to be configured.
Name	Name of the device.
State	The current status of the device.
Active/Actual Data	Only the settings of (a) RCM-24 module(s) will be shown.
Preset Data	Only the presets of (a) RCM-24 module(s) will be shown.
All Data	All parameters, i.e. settings and presets, of (a) RCM-24 module(s) will be shown.
SAVE DATA TO DEVICES	Writes the selected parameters into the RCM-24 module(s).
LOAD DATA FROM DEVICES	Reads the selected parameters out of the RCM-24 module(s).

COMPARE IRIS/DEVICE DATA	Compares the settings of selected data in IRIS-Net and the RCM-24 modue(s).
SHOW DIFFERENCES	Displays the differences in the parameters found during the comparison.

TESTING THE CONNECTED LOUDSPEAKER SYSTEMS

IRIS-Net allows convenient analysis of all loudspeaker systems connected to remote amplifiers used in a project. All remote amplifiers are listed in the RCM-24 System Check Window.

Loudspeaker system checks can be performed for various power amplifiers at the same time.



Element	**Description**
POWER ON	Takes the power amps selected in the list out of Standby Mode.
POWER OFF	Puts the amplifiers selected in the list into Standby Mode.
START TEST	Starts testing loudspeaker systems that are connected to power amps selected in the list of amplifiers.
SET REFERENCE	The measuring data of the previous test are stored as reference.
Hi	The level at which the loudspeaker systems test is performed.
SELECT ALL	Selects all amplifiers in the amplifier list.
CLEAR ALL	Deselects all amplifiers in the amplifier list.
EXTRACT MEASURE FILES	The measurement of the previous test is stored into a file.

Amplifier	Name of the amplifier consisting of CAN address, internal IRIS-Net name, and type of amplifier.
Amplifier Status	The current status of the amplifier.
Ch. A Speaker Status	The current status of the amplifier's channel "A".
Ch. B Speaker Status	The current status of the amplifier's channel "B".

OVERVIEW OF ALL AMPS USED IN A PROJECT

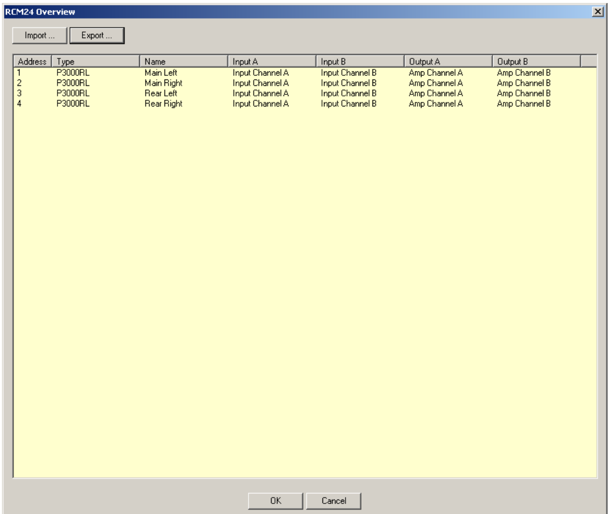
The Overview Window shows all remote amplifiers of the current project. Names can be assigned to amplifiers as well as to their input and output channels. Comfortable assignment of names in projects that include a large number of amplifiers is possible through exporting a structured text file.

This text file can be edited using basically any text editor application. The modified text file can be imported into IRIS-Net and the names included in the file are automatically assigned to the corresponding amplifiers as well as to their respective input and output channels.

Caution!

Using * (asterisk) and/or = (equal) signs in a name is not permissible.

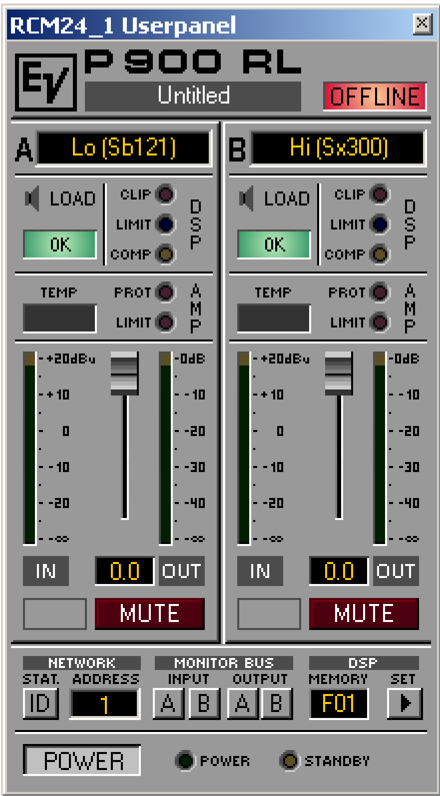
Consequences



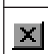
Element	**Description**
Import ...	Opens a text file containing the names of amplifiers including their input and output channels that has previously been exported (and edited).
Export ...	Stores a text file containing the names of amplifiers including their input and output channels.
Address	The CAN-BUS address of a remote amplifier.
Type	The remote amplifier model.
Name	The name of a remote amplifier.
Input A	The name of input channel "A".
Input B	The name of input channel "B".
Output A	The name of output channel "A".
Output B	The name of output channel "B".

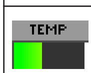
Amplifier Control Panel

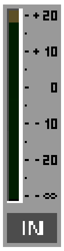
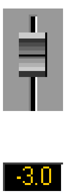
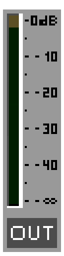
Double clicking with the left mouse button on an amplifier gets you to the Amplifier Control Panel, which provides access to the most important controls and indications of the selected amplifier.



Simultaneously opening several Amplifier Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.

Element	Description
P 900 RL	Amplifier Type (generated during amplifier selection or read from the amp while being on-line)
	Using the left mouse button, click on the Close button to close the Amplifier Control Panel.
Stage Left	A name can be assigned to each amplifier to specify its use or position. Click on the gray-shaded entry field below the Amplifier Type field and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: Entering amplifier names is also possible with in the Setup & Control Panel on the Config & Info page.**

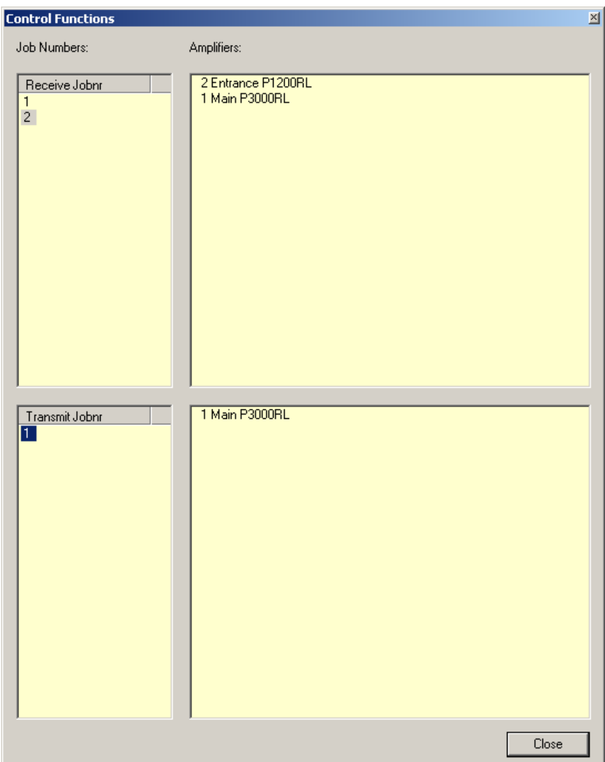
	The Online / Offline indicator signals whether the selected amplifier is included in the network or off-line. The red OFFLINE indicator signals that the corresponding amplifier is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding amplifier is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
	The amplifier channels are named channel A and B. A name can be assigned to each channel to easily identify its allocation and use. Using the left mouse button, click in the entry field and enter the desired name for the channel. Press Return on the keyboard to acknowledge your entry. **HINT: Entering channel names is also possible within the Setup & Control Panel on the Config & Info page.**
	The CLIP indicator lights whenever the signal of the internal signal processor clips. The signal processor's headroom is 12 dB, which is no problem when using normal filter settings. However, when drastically increasing the level of several adjacent or overlapping filters, distortion of high-level signals may occur, which the CLIP indicator indicates. In that case reducing the signal-level or trying a bit more moderate equalizer setting is recommended.
	The LIMIT indicator lights whenever the digital limiter of the corresponding channel is activated, e.g. when the signal level exceeds the specified threshold and the output level is being limited to this value.
	The COMP indicator lights when the digital compressor of the corresponding channel is activated, e.g. when the signal level exceeds the specified threshold and the output level being reduced.
	The LOAD indicator shows whether the load connected to the amplifier output is within the allowable range or if short-circuit or line interruption has occurred. The green OK-indication signals that the connected load is between the specified lower and upper limit values. These values are set in the Setup & Control Panel in the Load screen. The red OPEN indication signals line interruption. It lights whenever the connected load exceeds the upper limit value. The red SHORTED indication signals short-circuit at the amplifier output. It lights whenever the connected load falls below the lower limit value. **HINT: The connected load is monitored continually as soon as a signal with a voltage of > 250 m V is present at the output. Calculation of signal levels below that threshold is not possible and the indicator shows the last acquired state.**
	The TEMP display shows the amplifier's internal temperature as a graph. The indicator lights green whenever the amplifier is operated in its normal operational temperature range. The indicator lights yellow whenever the amplifier builds up heat because of continuous high output. However, since the internal fans provide sufficient ventilation there is no risk of thermal overload in this state. As soon as temperature indication changes to red, reducing the output level is strongly recommended. Otherwise the amplifier might cease operation because of thermal overload.
	When the red PROTECT indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated. In that case, the amplifier is separated via output relay from the connected load to prevent possible damage of the loudspeaker systems or the amplifier itself. The cause for the fault, a short-circuited speaker line for instance, must be remedied. In case of thermal over- load, you have to wait some time until the amplifier automatically re-enters normal operation. (see also Supervision & Test for additional fault information)

	The LIMIT indicator lights as soon as the internal dynamic limiter is activated, which is the case when the amplifier is operated at maximum out- put. Short-term blinking is not a problem, since the internal limiter controls input levels of up to +20dBu down to a distortion rate of approximately 1%. However, if this indicator lights permanently, reducing the output level is strongly recommended to protect the connected loudspeaker systems from being damaged by capacity overload.
	The Input Level Meters provide indication of the corresponding audio levels at the amplifier inputs in dBu. The amplifier's nominal input level is +6dBu, the maximum level can be as high as +21dBu. In general, it is recommended that the amplifier be operated in a range between 0 and +10dBu. Only signal peaks should be at higher levels.
	The level controls are for adjusting the overall amplification of the corresponding amplifier channel. Setting the level controls to a value between 0dB and -6dB provides full output capacity. The numerical field below the level controls indicates the set level, by which the output amplification is attenuated, in dB.
	The Output Level Meters provide indication of the corresponding audio levels at the amplifier outputs. Indication in dB is relative to amplifier full-modulation. A 0dB output level (full-modulation) is indicated in yellow.
	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Clicking this switch activates the STATUS indicator on the amplifier's rear panel as well as in the amplifier's front panel window in the IRIS-Net software. Normally, the STATUS indicator blinks only during serial communication. Once the STATUS switch is engaged, the STATUS indicator blinks in a steady but fast sequence. This function is meant for checking communication and for identifying or searching an amplifier in a large system setup.
	The address field indicates the set amplifier address. Assigning a new address is also possible by clicking into the field with the left mouse but- ton and entering the desired amplifier address. Available values are 1 to 250. Press Return on the computer keyboard to acknowledge your entry. The assigned address and the address specified by the setting of the selection switch on the amplifier's rear panel have to be identical. Each address can exist only once within a system.

	These buttons allow assigning amplifier channels to the monitor bus. The monitor bus allows monitoring any amplifier input or output signals within an installation. INPUT A / B selects the corresponding input signal while OUTPUT A / B allows switching between the output signals of channels A and B. Simply click on an amp channel's icon to select it for monitoring. The corresponding channel is assigned to the monitor bus. Any previous selection is simultaneously canceled, so that only the actually selected amp channel can be monitored. Clicking the button of an active amp channel separates the channel from the monitor bus.
	This field indicates the active factory or user preset. Each remote amp has a factory setting F01 offering linear settings and eight user-programmable presets U0...U08 for storing random user data. Loading and saving presets is done in the Setup & Control window.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all amplifier- and DSP-parameters, control and monitoring functions plus additional function groups.
	This soft-key allows switching an amplifier on or off. The STANDBY and POWER indicators signal the actual operational status. The Config & Info window allows programming individual power-on delays for each amplifier. Note: The power-on delay defaults to „Address * 150 ms". For address 8 the power-on delay default would be for example: 8 * 150 ms = 1200 ms.
	These indicators show the amp's actual operational status. STANDBY lights whenever the amplifier is in stand-by mode. POWER lights whenever the amplifier is powered-on and ready for operation. If neither one of the indicators lights, the amplifier is either off-line or powered-off.

Control Functions

Communication amongst amplifiers is possible through sending and receiving messages that are referred to as "job codes". In principle, a job code is a function number. Amplifiers send these function numbers via the CAN bus. A single or several amplifiers are able to receive and interpret these job codes and execute the associated functions. Each amplifier can send and receive up to 5 different job codes.
The Control Functions Window lists the job code numbers that are used by all the amplifiers in a project like they were defined in the amplifiers' Config & Info Window. When clicking on a number in the list of job code numbers in the left part of the window, all amplifiers using this number are listed in the right part of the window. This, for example, allows determination whether a specific job code number has been configured for all amplifiers in a project.



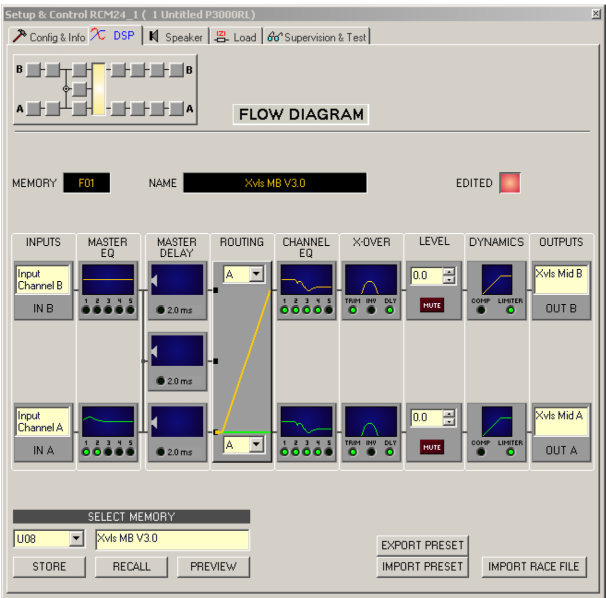
Setup & Control

The Setup & Control window allows configuring all amplifier parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

Window	Description
Config & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Filter, Delay, X-Over, Dynamics) of the amplifier.
Speaker	This page allows loading and displaying speaker data.

Load	This page provides access to several settings for impedance/load monitoring and impedance testing.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.

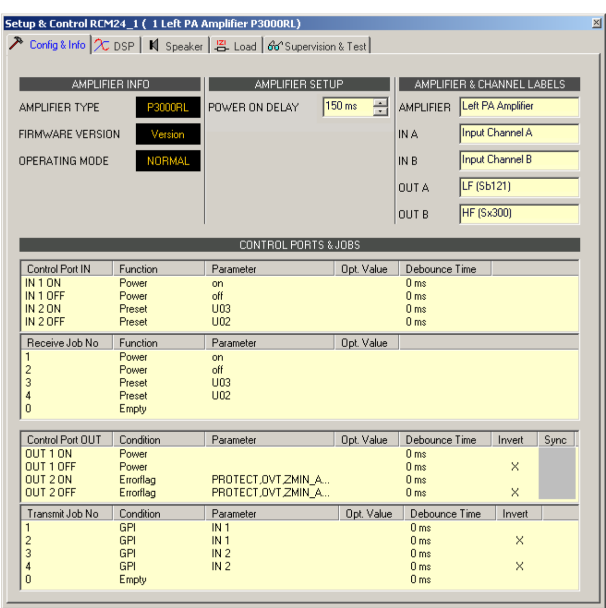
Clicking on the soft key SET in the Amplifier Control Panel opens the Setup & Control window.



CONFIG & INFO

The Config & Info window provides information and basic settings for the selected amplifier. Additionally, editing labels and configuring control functions is possible as well.

To select the page click onto the Configuration & Information register in the Setup & Control Window.



Element	Default	Range	Description
AMPLIFIER TYPE			Shows the amplifier type
FIRMWARE VERSION			Shows the amp software's version number (operating system, firmware)
OPERATING MODE			Shows the amplifier's operating mode (only with P3000RL). The P3000RL can be operated in NORMAL or BRIDGED mode.
POWER ON DELAY	Address * 150 ms	50...4000 ms 50 ms Steps	Allows programming an amplifier's power-on delay. Setting different delay times is recommended to prevent the mains fuse from blowing when powering on several amps at the same time. The value defaults to address * 150 ms
AMPLIFIER & CHANNEL LABELS AMPLIFIER Left PA Amplifier IN A Input Channel A IN B Input Channel B OUT A LF (Sb121) OUT B HF (Sx300)			The labels of an amplifier and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately adopted in the different panels and windows (amplifier control panel, flow diagram, overview).

A control port offering two control inputs and two control outputs is located on the amplifier's rear panel. The functions of these inputs and outputs can be randomly programmed. For example, the control inputs (GPI) can be used for power- on / stand-by or preset switching as well as for changing parameters. The control outputs (GPO) are for signaling internal statuses. They can directly trigger LEDs, control lamps or relays. In the Supervision & Test window the states of the control inputs are displayed and you have the possibility to switch the control outputs manually. For more information and electrical specifications of the control port, please refer to the amplifier manuals.

Control Inputs: Each status change of a control input can trigger a function. Different functions can be assigned for the opening (OFF) or closing (ON) of a contact.

Example:

Control Port IN	Function	Parameter	Opt. Value	Debounce Time
IN 1 ON	Power	on		0 ms
IN 1 OFF	Power	off		0 ms
IN 2 ON	Preset	U03		0 ms
IN 2 OFF	Preset	U02		0 ms

This example shows the programming of two control inputs where IN1 switches the amplifier on or off and IN2 selects presets U02 or U03.

- IN1 ON: Power on (closing the contact of control input 1 switches the amplifier on)
- IN1 OFF: Power off (opening the contact of control input 1 switches the amplifier to stand-by)
- IN2 ON: Preset U03 (closing the contact of control input 2 selects preset U03)
- IN2 OFF: Preset U02 (opening the contact of control input 2 selects preset U02)

Element	Default	Range	Description
Control Port IN		IN 1 ON IN 1 OFF IN 2 ON IN 2 OFF	This provides a listing of the two control inputs and their statuses ON and OFF. The entries in the corresponding lines specify the action when closing (ON) or opening (OFF) a contact.
Function	(empty)		This column allows assigning functions to a control input's statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. Following a status change the assigned function is initiated after the set time interval has past.

Control Outputs: Internal status changes inside of the amplifier, like for example operational faults, alerts when exceeding parameter limits, and internal operational statuses can be signaled to external systems or central control units.

Example:

Control Port OUT	Condition	Parameter	Opt. Value	Debounce Time	Invert	Sync
OUT 1 ON	Power			0 ms		
OUT 1 OFF	Power			0 ms	X	
OUT 2 ON	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms		
OUT 2 OFF	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms	X	

This example shows the programming of two control outputs where OUT1 signals whether the amplifier's power is switched ON or OFF while OUT2 signals faulty operation.

- OUT1 ON: Power (control output 1 is closed when the amplifier's power is switched on)
- OUT1 OFF: Invert Power (control output 1 is open when the amplifier's power is switched off / stand-by mode)
- OUT2 ON: Errorflag (control output 2 is closed when operational faults according to the parameter list have occurred)
- OUT2 OFF: Invert Errorflag (control output 2 is open when no faults have occurred)

Element	Default	Range	Description
Control Port OUT	0	OUT 1 ON OUT 1 OFF OUT 2 ON OUT 2 OFF	This provides a listing of the two control outputs and their statuses ON and OFF. The entries in the corresponding lines specify which status results in the closing (ON) or opening (OFF) of a contact.

Condition	(empty)		This column allows assigning internal events (conditions) to a control output's statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. An event is signaled following an internal status change and after the specified time interval has past.
Invert	(empty)	(empty) / X	This column allows entering whether a status is signaled when the specified Condition is „true“ (no entry) or „false“ (click „X“ to signal an inverted state).
Sync	(empty)		This column displays the SYNC flag. „X“ specifies that the output is synchronized with a sync-signal. This flag is erased when entering a new Function.

Jobs

For amplifiers to be able to communicate with each other, it is possible to send and receive Job Codes. In principle, a job code is a function number that an amplifier transmits via CAN-bus and that is received and interpreted by another or several other amplifiers. Each amplifier is capable of transmitting and receiving up to 5 different job codes.
Programming job codes is nearly identical to the programming of control inputs and outputs.
Receive Jobs: A receive job is a function that is carried out as soon as the corresponding function number (the Receive Job Code) is received.
Example:

Receive Job No	Function	Parameter	Opt. Value
1	Power	on	
2	Power	off	
3	Preset	U03	
4	Preset	U02	
0	Empty		

This example shows the programming of four Receive Jobs. Jobs No. 1 and 2 switch the amplifier's power on or off while jobs No. 3 and 4 select presets U03 or U02. The fifth Receive Job has not been configured.

– Receive Job Nr. 1: Power on (receiving Job Code 1 switches the amplifier's power on)
– Receive Job Nr. 2: Power off (receiving Job Code 2 switches the amplifier into stand-by mode)
– Receive Job Nr. 3: Preset U03 (receiving Job Code 3 selects preset U03)
– Receive Job Nr. 4: Preset U02 (receiving Job Code 4 selects preset U02)

Element	**Default**	**Range**	**Description**
Receive Job No	0	1...1023	Here, you can specify which incoming job code numbers a specific amplifier recognizes. Entering random numbers between 0 and 1023 is possible.

Function	(empty)		This column allows assigning an individual function to each job code received. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.

HINT: Programming identical control functions or receive jobs for several amps is easily accomplished by creating a group that includes all the desired amps and afterwards perform the programming in the group's Configuration & Information dialog. All settings are automatically applied to all amplifiers of that group, which saves time and effort and additionally reduces the risk of programming errors.

Transmit Jobs: Transmit Job defines a function number that is sent as soon as a specific internal event (condition) occurs in the amplifier.

Example:

Transmit Job No	Condition	Parameter	Opt. Value	Debounce Time	Invert
1	GPI	IN1		0 ms	
2	GPI	IN1		0 ms	X
3	GPI	IN2		0 ms	
4	GPI	IN2		0 ms	X
0	Empty			0 ms	

This example shows the programming of four Transmit Jobs. Jobs No. 1 and 2 are triggered by control input 1. Jobs No. 3 and 4 are triggered by the status signaled from control input 2. The fifth transmit job has not been configured.

- Transmit Job Nr. 1: GPI IN1 (Job Code 1 is transmitted when control input 1 is closing)
- Transmit Job Nr. 2: Invert GPI IN1 (Job Code 2 is transmitted when control input 1 opens)
- Transmit Job Nr. 3: GPI IN2 (Job Code 3 is transmitted when control input 2 is closing)
- Transmit Job Nr. 4: Invert GPI IN2 (Job Code 4 is transmitted when control input 2 opens)

Element	Default	Range	Description
Transmit Job No	0	1...1023	Here, you can specify which job code numbers an amplifier transmits on the occurrence of specific events. Entering random numbers between 0 and 1023 is possible.
Condition	(empty)		This column allows specifying an event (condition) that triggers the corresponding transmit job code. Clicking the desired line in the Condition menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.

Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here, you can program delay or debouncing times. A transmit job code is sent following a specific event and after the specified time interval has past.
Invert			This column allows entering whether a job code is transmitted when the specified Condition is "true" (no entry) or "false" (click "X" to signal an inverted state).

Input and Receive Job functions: The following table lists all functions together with their individual settings, which can be triggered via control input or Receive Job.

Function	Parameter	Opt. Value	Function executed
Empty	-	-	None
Power	off on flip		Power Off (Standby) Power On Power-status change (ON to Stand-by and reverse)
Absolute	All DSP parameters	Corresponding Parameter Value (parameter-dependent)	Set the specified absolute parameter value for the selected parameter
Relative	All DSP parameters	Parameter Value Off- set (parameter-dependent)	Changes the actual value of the selected parameter by the specified offset value
Flip	Parameters with two statuses		Changes the status of the selected parameter (e.g. bypass On / Off)
Preset	U01 - U08, F01		Changes a preset to the specified preset number
Monitor	Relay, IN A, IN B, OUT A, OUT B	on, off	Activates respectively deactivates the selected monitor bus signal
Ground fault	A, B		Resets the ground-fault error flag of selected amplifier channels
Memo flag	Set, Clear, Toggle Memo flags 1 - 16		Sets, erases or changes selected memory flags. Up to 16 memory flags are available and simultaneously accessible.
Measurem ent	Generator frequency, Time, Level A / B		Starts the test generator with a tone signal of the specified frequency at the levels specified for channels A / B for the selected duration (0 ms = infinite)

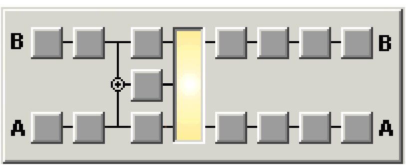
Output and Transmit Job Conditions: The following table lists all amplifier statuses that can be used for triggering control outputs or for sending Transmit Job Codes.

Function	Parameter	Opt.Value	Invert	Triggering Event/Status Change
Empty	-	-		Not configured
Power			 X	Power On Power Off (Standby)
Absolute	all DSP parameters	Corresponding Parameter Value (parameter-dependent)	X	Set parameter value reached or exceeded Set parameter value declined
Temp	Temperature in °C		 X	Set temperature reached or exceeded Set temperature declined
VU	IN A, IN B, OUT A, OUT B, Amp Limiter A/B, DSP Limiter A/B, Compressor A/B	Level in dB	X	Set level reached or exceeded Set level declined
GPI	IN 1, IN 2		 X	Control input 1 / 2 closed (ON) Control input 1 / 2 open (OFF)
Errorflag	All internal fault conditions		X	Single or several error flags set None of the selected error flags set
Memoflag	Enable for selected flags as well as bit-pattern of flags 1 - 16		X	Memory flags match the selected bit-pattern Memory flags do not match the selected bit-pattern
Preset	U01 - U08, F01		 X	Specified preset selected Other than the specified preset selected

DSP

The DSP pages provide overview and access to all DSP parameters of an amplifier. Within this window you can use the Flow Diagram Selector to link to different function groups.

FLOW DIAGRAM SELECTOR
The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.

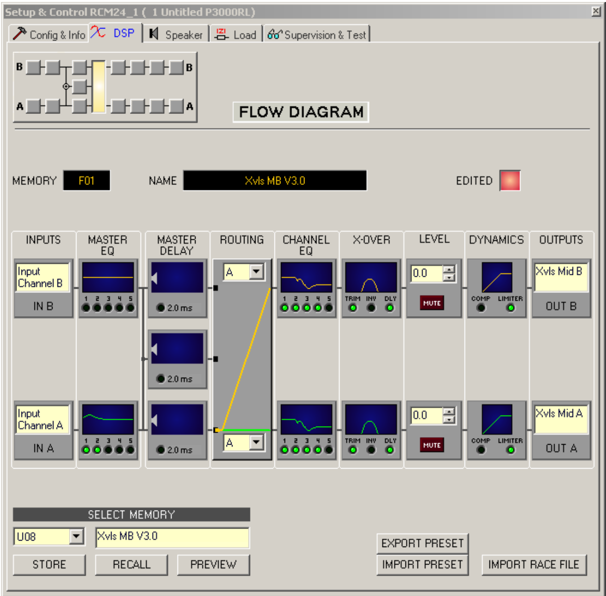


A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

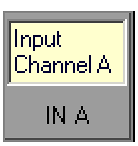
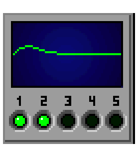
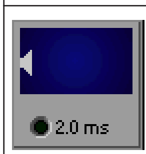
Page	Description
Flow Diagram	The signal flow display provides an overview of an amplifier's DSP settings. This area also includes all controls for the preset location and preset file management.
Master EQ	The MASTER EQ page provides access to the two 5-band parametric equalizers of the amplifier inputs.
Master Delay	This page allows the programming of delay lines for the amplifier channels A and B as well as for the summed input A+B.
Channel EQ	The CHANNEL EQ page offers access to the two 5-band parametric equalizers of the amplifier outputs for speaker equalization.
X-Over	Frequency crossover-filters as well as the parameters gain, polarity and alignment-delay for both channels are located in the X-OVER area.
Dynamics	This page provides access to compressor and limiter of each amplifier channel.

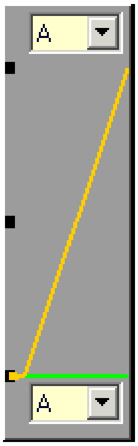
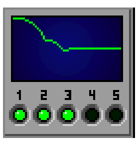
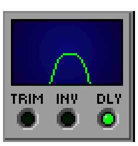
The DSP functions of a remote amplifier can be accessed by clicking onto the SET key in the Amplifier Control Panel followed by a click on the DSP register in the Setup & Control Window.

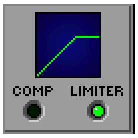
FLOW DIAGRAM
The FLOW DIAGRAM window shows a signal flow diagram, which offers a quick overview of all DSP setting of an amplifier. Labeling and routing channels can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for the saving, loading and previewing of loudspeaker presets are also accessible from this window.
The FLOW DIAGRAM window opens when clicking on the first, fourth or eighth block in the Flow Diagram Selector.



Function Blocks

Element	Description
	Input Block: The text field allows specifying a name for the corresponding input channel. A click with the right mouse button onto IN A or IN B opens the Copy & Paste menu, which allows copying all parameters of the corresponding input channel (Master EQ, Master Delay) to any other input channel within the same project.
	Master EQ Block: The Master EQ block displays the 5 Master EQs of the corresponding input channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Master EQ block. A single click with the left mouse button onto this block opens the MASTER EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Master Delay Block: This displays the Master Delay of the input channels. The corresponding LED signals whether a delay has been programmed or not. The delay-value is displayed together with the measurement unit next to the LED. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the MASTER DELAY page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Master Delay block to any other Master Delay block within the same project.

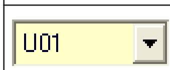
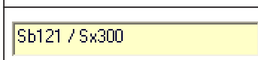
	Routing Block: Here you can assign the output channel routing. The selection can be performed using the two combination boxes or directly in the graphic display by grabbing the left ends of the yellow respectively green lines with the mouse and dragging and dropping them over the desired input channel. A click with the right mouse button onto routing block opens the Copy & Paste menu of all DSP settings, which allows copying all DSP parameters of an amplifier to any other amplifier within the same project.
	Channel EQ Block: The Channel EQ block displays the 5 Channel EQs of the corresponding output channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Crossover Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse button onto this block opens the X- OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X- Over block within the same project.
	Level Block: The numerical field is identical to the numerical field below the level controls in the Amplifier Control Panel. So the field indicates the actually set attenuation, by which the internally specified amplification is attenuated, in dB. The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.

	Dynamics Block: This block provides graphical display of the dynamics functions of the corresponding output. The two LEDs indicate whether compressor or limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Dynamics block to any other Dynamics block within the same project.
	Output Block: The text field allows specifying a name for the corresponding output channel. A click with the right mouse button onto OUT A or OUT B opens the Copy & Paste menu, which allows copying all parameters of the corresponding output channel (Routing, Channel EQ, X-Over, Dynamics) to any other output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied.

Status Indication

Element	Description
	The MEMORY display shows the number according to the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	NAME indicates the name of the actually audible preset.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

Store / Recall / Preview

Element	Description
	Lets you select a preset number. The selection is valid for all following actions, e.g. RECALL, PREVIEW or STORE.
	This field allows assigning a name to a preset before saving it. The name is stored as well and is displayed in the status line NAME after performing a RECALL.
	STORE saves all momentary set DSP parameters together with the entered name into the specified preset.

RECALL	RECALL loads and displays all DSP parameters that are stored in the selected preset. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
PREVIEW	PREVIEW reads and displays all DSP parameters that are stored in the selected preset. This function is used to display and check a preset’s contents, without actually loading the preset. You can neither listen to the preset nor edit its contents, as long as you do not explicitly load it using the RECALL function.

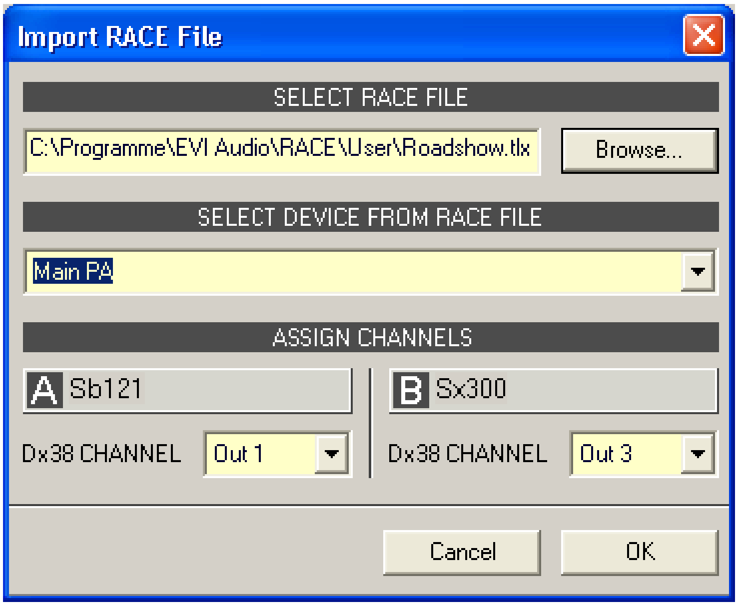
Import / Export of Preset Files

IRIS-Net allows the storing of all DSP parameters of an amplifier together with the according preset name in a file, and to load amplifier parameters from these files. Therefore, IRIS-Net creates a sub-directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory /Presets is permissible.

Element	Description
IMPORT PRESET	After clicking onto IMPORT PRESET appears an Open File dialog box. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in online mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loud speaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a Save File dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store all DSP parameters together with the corresponding file name. “.ds” is automatically added as file extension.

Import of EV RACE Files

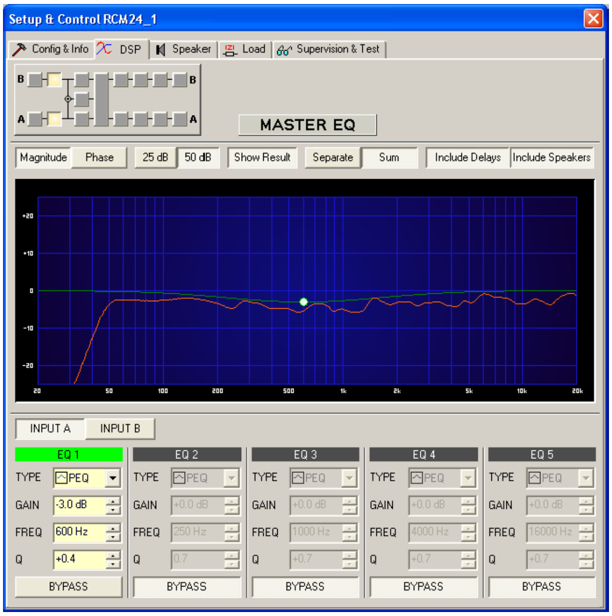
Element	Description
IMPORT RACE FILE	IRIS-Net allows importing loudspeaker presets that have been created in Electro-Voice RACE by clicking onto the button IMPORT RACE FILE to open the following dialog box.



First, you have to select the desired RACE file by use of the Browse... button. Because a RACE file can hold the data of up to 31 EV Dx38, you need to continue by selecting the desired device from the RACE file within the dialog SELECT DEVICE FROM RACE FILE. At the end you have to specify which of the four Dx38 output channels should be assigned to the corresponding amplifier channels. Clicking onto OK button completes the process.

Caution!
The loaded RACE file becomes instantly audible when in on-line mode. Be sure to select the desired file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

MASTER EQ
Both input channels of a remote amplifier employ 5-band parametric equalizers each, which allow programming highly variable full-range speaker equalization to match a PA-system to different environmental and acoustical requirements. In many cases post-mixing console parametric equalization becomes redundant.
The Master-EQ is selected by clicking on the second block of the flow diagram selector or by double clicking on the MASTER EQ block in the full-scale flow diagram.



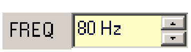
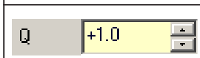
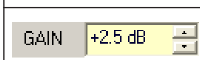
Graphics Display Indication

Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings – the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while “electrical” graphs are indicated in dark colors.
Separate Sum	Selecting “separate” results in a separated display of the two amplifier channels’ transfer functions while “sum” shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the “Speaker” register sheet.

Channel Selection

Element	Description
INPUT A INPUT B	Switch for selecting input A or input B for filter editing. A click with the right mouse button opens the “Copy & Paste” menu, which allows convenient copying all EQs of the corresponding input to any other EQ-filter bank within the same project.

Filter Parameters

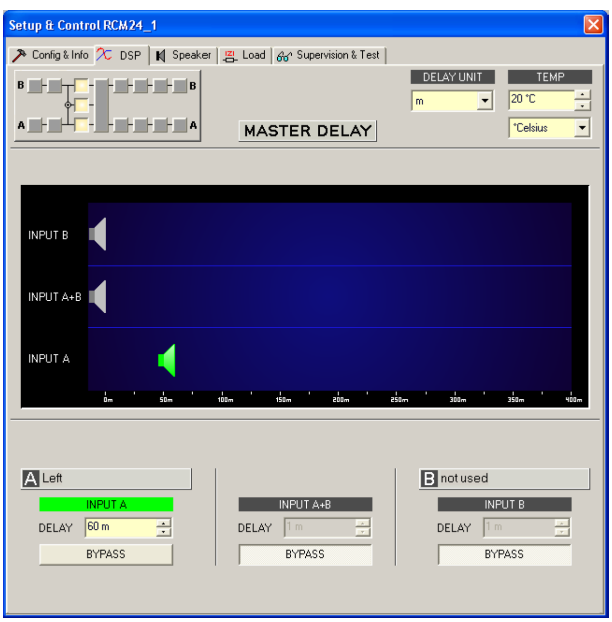
Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the “Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following edit- able parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
	63 / 250 / 1000 /	20 Hz...20	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving
	4000 / 16000 Hz	kHz	and Hi / Lo pass filters.
	0.7	0.4...40.0	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		(PEQ),	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the res-
		0.4...2.0 (Hi-/	ponse of Hi, Lo and All pass filters with slopes of 12dB/oct.
		Lopass)	
	0 dB	-18...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via “Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its amplification (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

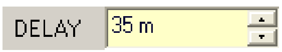
MASTER DELAY

Individual master delays can be set for each input channel of a remote amplifier. Setting a different delay for the summed signal of the two input channels is also possible. Master Delays are mainly used to compensate for different natural delay times in the audio signal, as they are common when two sound sources reproducing identical audio information are located further apart.
You can select the master delay window by clicking onto the third block in the Flow Diagram Selector or by double clicking onto the MASTER DELAY block in the flow diagram.

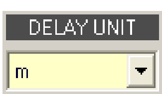
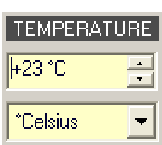


Channel Parameters

Element	Default	Range	Description
			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all master delay parameters of the selected input to any other master delay within the same project.

			Channel identification A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all master delay parameters of the selected input to any other master delay within the same project.
	2.0 ms	2.0...1000 ms	DELAY allows delaying the corresponding input channel's audio signal by an adjustable period of time. **HINT: The amplifier accepts settings up to a total delay time of 1365ms per channel, including Master Delay and X-OVER Delay. If a comparably long alignment delay has already been set for the X-OVER, the delay time that is available for the master delay scan fall below 1000ms!**
			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

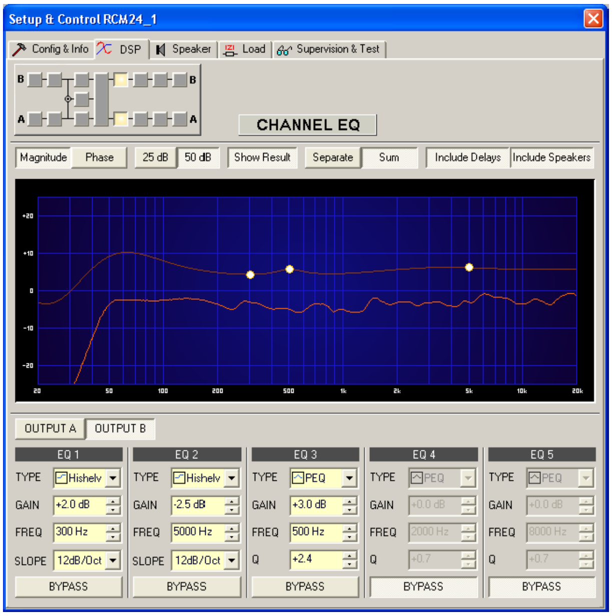
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 °C	-20...60 °C or -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by Dragging the Mouse in the Graphics Display

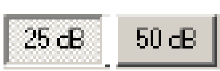
The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel's delay time. A delay's title lights in color as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

CHANNEL EQ

Both output channels of a remote amplifier employ 5-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select All pass as filter type, these filters are identical to the ones of the master-EQ's.
The Channel-EQ is selected by clicking on the fifth block of the flow diagram selector or by double clicking on the CHANNEL EQ block in the full-scale flow diagram.



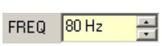
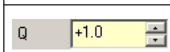
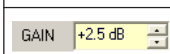
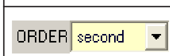
Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
	Selecting separate results in a separated display of the two amplifier channels' transfer functions while sum shows the summed signal of the amplifier channels.
	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

Channel Selection

Element	Description
	Switch for selecting output A or output B for filter editing. A click with the right mouse button opens the Copy & Paste menu, which allows convenient copying all EQs of the corresponding output to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the „Copy & Paste“ menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
	63 / 250 / 1000 / 4000 / 16000 Hz	20 Hz...20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
	0.7	0.4...40.0 (PEQ), 0.4...2.0 (Hi-/ Lopass), 0.4...2.0 (Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct.
			GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with All pass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via Mouse Movement in the Graphics Display

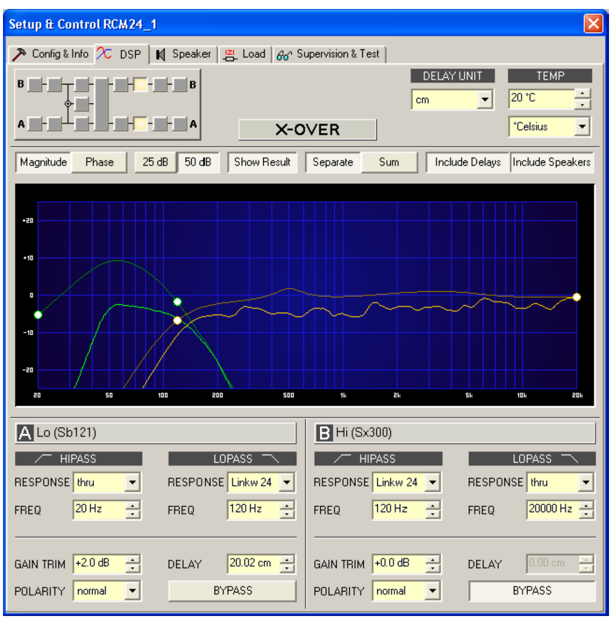
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter's frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

X-OVER

The X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch, which are provided for each output channel of a remote amplifier. By means of these parameters you are able to correctly configure a multi-way speaker system's individual frequency bands, compensate for natural delays and adjust levels.

Clicking on the sixth block in the Flow Diagram Selector or double clicking on the X-OVER block in the large signal flow diagram opens the X-Over window.



Graphics Display Indication

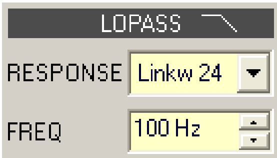
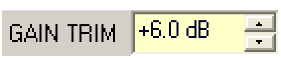
The graphics display offers several different indication modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (Master EQ, Channel EQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
Separate Sum	Selecting separate results in a separated display of the two amplifier channels' transfer functions while sum shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

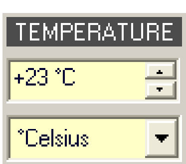
Channel Parameters

Element	**Default**	**Range**	**Description**
A Sb121			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all X-Over parameters of the corresponding out- put to any other X-Over within the same project.
HIPASS RESPONSE Linkw 24 FREQ 100 Hz	thru, 20 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butter-worth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI- PASS filter to any HI-PASS filters within the same project.

	thru, 20000 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO- PASS filter to any LO-PASS filters within the same project.
	0 dB	-30 dB...6 dB	GAIN TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
	normal	normal, inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to Sum).

	0.0 ms	0.0...1000.0 ms	DELAY allows delaying the audio signal of the corresponding output by an adjustable period of time. This delay method is typically used as time-alignment-delay to overcome negative sound effects like they result from different distances between loudspeaker systems within one cabinet or the positioning of speakers in a PA- installation that otherwise would cause a high amount of natural delay. **HINT: The amplifier accepts settings up to a total delay time of 1365 ms per channel including MASTER DELAY and X-OVER DELAY. If a comparably long MASTER DELAY has already been set, the remaining delay time that is available for the output delay can fall below 1000ms!**
			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

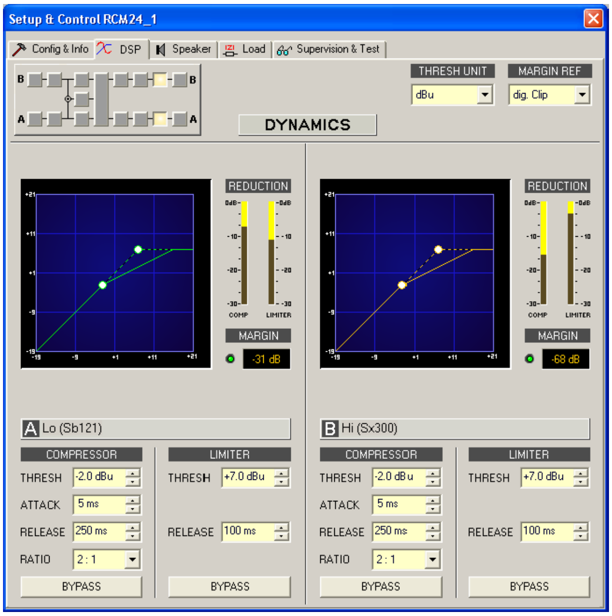
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 °C	-20...60 °C or -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing X-Over Filters by Dragging the Mouse in the Graphics Display

Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right.
A filter's title lights in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling. An extra white graph is displayed in addition, representing the frequency response of the corresponding selected filter.

DYNAMICS

Each output channel of a remote amplifier offers a compressor and a limiter. These functions can be accessed via the dynamics window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.
Clicking onto the seventh block in the Flow Diagram Selector or double clicking onto the DYNAMICS block in the large flow diagram opens the dynamics window.



Channel Parameters

Element	Description
A Sb121	Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all dynamics parameters of the corresponding channel to any other channels within the same project.

Compressor Parameters

Element	Default	Range	Description
COMPRESSOR			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all compressor parameters of the corresponding channel to any other channels within the same project.
THRESH -3.0 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	The THRESHOLD parameter determines the audio signal level above which the compressor starts operating.
ATTACK 5 ms	5 ms	0...99 ms	ATTACK determines how fast the compressor reduces amplification when the threshold is exceeded.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the compressor returns to normal amplification, after the audio signal level declined the threshold.

RATIO 4 : 1	2:1	1 : 1, 1.4: 1, 2: 1, 4: 1, 8:1	RATIO determines the amount of compression that the audio signal is compressed when exceeding the threshold. The setting of 4 : 1 for example relates to a reduction of the audio signal by the factor 4.
BYPASS			BYPASS switches the compressor on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the compressed and non-compressed audio signals.

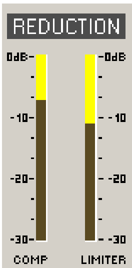
Limiter Parameters

Element	Default	Range	Description
LIMITER			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all limiter parameters of the corresponding channel to any other channels within the same project.
THRESH -3.0 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	THRESHOLD determines the audio signal level above which the limiter starts operating.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
BYPASS			BYPASS switches the limiter on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the limited and non-limited audio signals.

General Parameters

Element	Default	Range	Description
THRESH UNIT dBu	dBu	dBu / Volts	This lets you select the unit for the threshold parameter. The selected setting applies to compressor and limiter as well.
MARGIN REF dig. Clip	dig. Clip	dig. Clip, Limi- ter Thresh	This lets you set the absolute level for margin indication. You can select between Digital Clip (relates to +21 dBu) and Limiter Threshold. The margin level indicates the distance between signal level and the set absolute level. The displayed margin always relates to the highest actual signal level reading.

Indications

Element	Description
	These indicators show the reduction in dB that is applied to the audio signal by the compressor (COMP) or limiter. Level reduction is indicated as vertical yellow bar graph.
	The margin level indicates the distance between signal level and the set absolute level. The displayed margin relates to the highest actual signal level reading since the last reset of the indicator. The LED changes from green to red as soon as the signal level reaches or exceeds the set absolute level (Digital Clip / Limiter Threshold). A click with the right mouse button onto the margin level followed by click onto Reset re-sets indication.

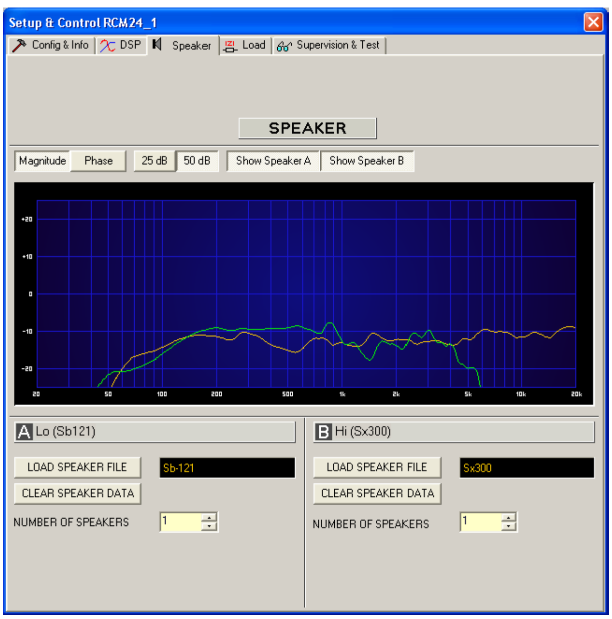
Editing Compressor / Limiter Parameters by Dragging the Mouse in the Graphics Display

Active compressors or limiters (bypass button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding compressor or limiter by vertically dragging the mouse. A click with the right mouse button onto the white dot of a compressor and keeping the mouse button pressed down lets you edit the ratio of compression.

A compressor's / limiter's title lights in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

Speaker

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the amplifier channels and display the acoustic results of this virtual combination. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of all common Electro-Voice loudspeaker systems. Some examples are provided in the IRIS-Net directory Speaker Files.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the amps. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of amplifier plus speaker transfer functions.
Clicking on the Speaker tab in the Setup & Control window opens the Speaker page.



Indication on the Graphic Display

Element	Description
Magnitude Phase	Switch for toggling between frequency response (magnitude) and phase response (phase) display
25 dB 50 dB	Switch for adjusting the scale of the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Speaker A Show Speaker B	Switching the display of the corresponding speaker data for an amplifier channel on/off is performed using the ”Show Speaker A” and ”Show Speaker B” switches.

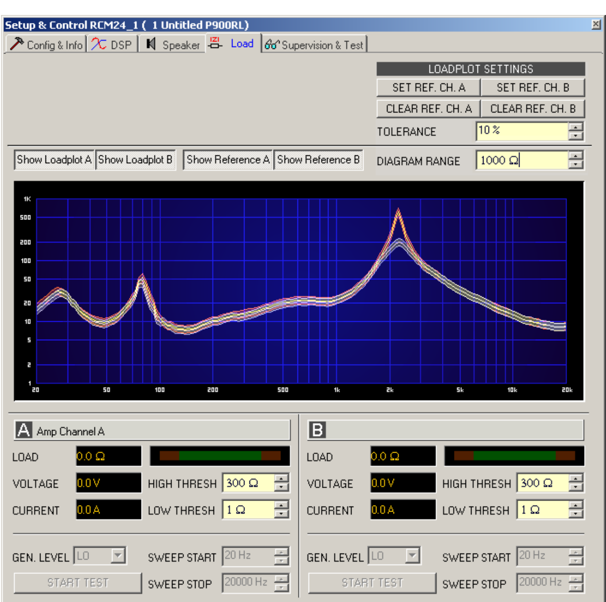
Channel Parameters

Element	Default	Range	Description
A Lo (Sb121)			Channel description and channel name

LOAD SPEAKER FILE Sb-121			Clicking the button LOAD SPEAKER FILE opens a dialog that allows the selection of the desired speaker file. Different loudspeaker files can be found in the Speaker files IRIS-Net folder. The name of the loaded loudspeaker model is shown in the black-shaded field on the right.
CLEAR SPEAKER DATA			Clicking the CLEAR SPEAKER DATA button clears the previously loaded measured speaker data of the selected channel.
NUMBER OF SPEAKERS 1	1	1...8	The NUMBER OF SPEAKERS parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel. Set- ting an amount from 1 to 8 is possible.

Load

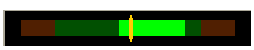
The Load window provides access to all settings and functions for testing and monitoring the load connected to the amplifier outputs.
The constantly measured output voltage and output current values of the Remote Power Amplifiers are indicated within the Load window. As soon as the output voltage of the signal present exceeds 150 mV, the resulting load is calculated and indicated. If the set thresholds are being exceeded or fallen short of, a corresponding message appears in the Load display of the Amplifier Control Panel. This dialog box permits to independently set the upper and lower thresholds for each power amp channel.
Within the Load window it is also possible to measure speaker impedance graphs and save them as references. The frequency range (start frequency, stop frequency) and the generator level of the sine-sweep test signal that is generated for this test can be adjusted. Specifying a tolerance field for the saved reference graphs is possible as well. A fault message is displayed in the event that a measurement exceeds or falls short of the tolerance range during system check. Select the Load window by clicking on the Load tab in the Setup & Control Window.



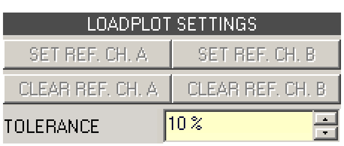
Indication on the Graphic Display

Element	Default	Range	Description
Show Loadplot A Show Loadplot B			The switches Show Load plot A and Show Load plot B turn the indication of the corresponding impedance graphs ON or OFF.
Show Reference A Show Reference B			The switches Show Reference A and Show Reference B turn the indication of the corresponding reference impedance graphs ON or OFF.
DIAGRAM RANGE 1000 Ω	1000 Ohm	50 Ohm...10 kOhm	DIAGRAM RANGE allows zooming in or out the diagram's impedance range (Y- axis).

Parameters And Indications For The Continuous Monitoring Of The Load Connected

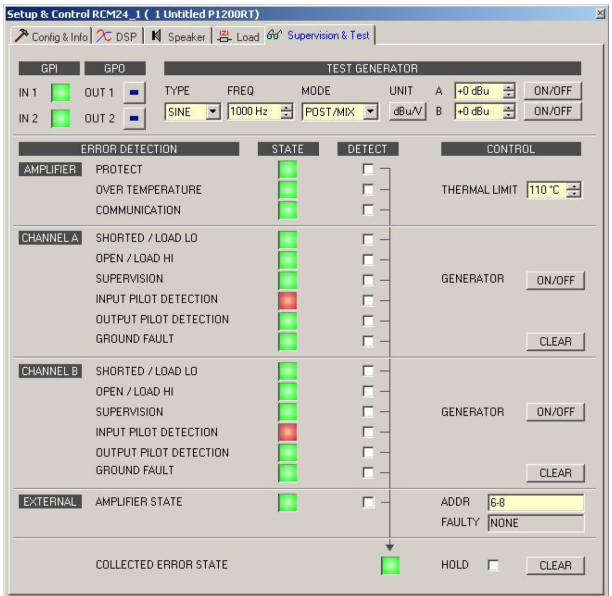
Element	Default	Range	Description
LOAD 24.8 Ω			The load display indicates the quotient of measured voltage and current (U/I).
			This indication shows the actual measured load, the progression, and the set value range. The orange needle indicates the actual value. The bright green bar indicates which loads have already been measured while being on-line. A red indication signals that the value exceeded or fell short of the set value range. The dark green area represents the allowable value range for the load of the corresponding power amp channel. The set HIGH THRESH respectively LOW THRESH values define the limits for this value range. Moving the cursor over the indication bar brings up a tool-tip context menu showing the numerical value of the lowest, the highest, and the actually measured load values. Clicking with the right mouse button on the indication bar, followed by a click on Reset, clears the previously measured load values (bright green and red ranges disappear).
VOLTAGE 0.0 V			The VOLTAGE display provides continuous indication of the corresponding power amp channel's output voltage.
CURRENT 0.0 A			The CURRENT display provides continuous indication of the corresponding power amp channel's output current.
HIGH THRESH 300 Ω	300 Ohm	0.0 Ohm...70 kOhm	HIGH THRESH sets the upper limit of the allowable impedance range (= minimum load). Once this value is exceeded, an OPEN fault message (line interrupt) appears in the Amplifier Control Panel.
LOW THRESH 1.0 Ω	1.0 Ohm	0.0 Ohm...70 kOhm	LOW THRESH sets the lower limit of the allowable impedance range (= maximum load). Once this value is fallen short of, a SHORTED fault message (line short-circuit) appears in the Amplifier Control Panel.

Parameters for Impedance Measurement

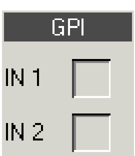
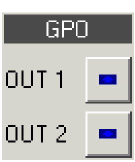
Element	Default	Range	Description
	LO	LO, HI	GEN. LEVEL sets the generator level for speaker impedance testing. The generator level LO corresponds to -10dBu (245 mV), the generator level HI corresponds to 0dBu (775 mV). **CAUTION: Extremely high levels during measurement may result in seriously damaging connected components.**
	20 Hz	20 Hz...20 kHz	SWEEP START sets the start frequency of the sine-sweep signal for speaker impedance testing.
	20 kHz	20 Hz...20 kHz	SWEEP STOP sets the stop frequency of the sine-sweep signal for speaker impedance testing.
			Clicking the START TEST soft-key launches the speaker impedance test. The generated sine-sweep signal sweeps over the previously defined frequency range. The graph of the measured impedance values is indicated in the Load plot display. Clicking this soft-key again cancels the test at any time.
	10%	5%...50%	Clicking the SET REF. CH. A and/or SET REF. CH. B soft-key saves the last test as reference. Clicking the CLEAR REF. CH. A and/or CLEAR REF. CH. B soft-key clears the corresponding reference. TOLERANCE defines the allowable deviation of the impedance graph. Actual measured test results and saved tolerance ranges are being compared during system check. A fault message is displayed if any point of the actual measurement falls outside of this tolerance range. The tolerance range is graphically displayed as spread in the corresponding color that surrounds the reference graph.

Supervision & Test

The Supervision & Test Dialog integrates functions for testing and monitoring power amps.
You can check control input states and trigger control outputs. A testing generator that provides sine, pink noise and white noise signal output allows acoustical testing. Status indicators for general power amp operation, the two amplifier channels and the load connected, indicate whether everything is okay or where failures occurred. You have the option to choose, which errors are combined and indicated in a general fault message.
A click on the Supervision & Test tab selects the page while in the Setup & Control Window.



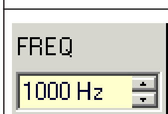
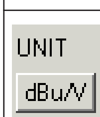
CONTROL INPUTS AND CONTROL OUTPUTS

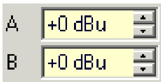
Element	Description
	This dialog indicates the actual states of the two freely programmable control inputs IN1 and IN2. A green indicator signals "not active", i.e. the control input is open or "high". A red indicator signals "active", in that case the control input is connected to the ground or "low".
	This dialog is for manually controlling the two Open Collector control outputs OUT1 and OUT2. Not engaged (blue) indicates that the control output is deactivated or highly resistive while engaged (red) indicates that the control output is activated and connected to the ground (closed). **HINT: When a control output has already been programmed, the programmed function defines the state of the control output and manual control is not possible.**

For detailed explanation on how to program control inputs and outputs please refer to chapter Config & Info.

TEST GENERATOR PARAMETERS

The test generator allows outputting a selected test-tone at an adjustable level via the power amps channel A and/or channel B which allows testing the cable run from the amplifier output to the connected loudspeaker systems as well as testing the functionality of the loudspeaker components.

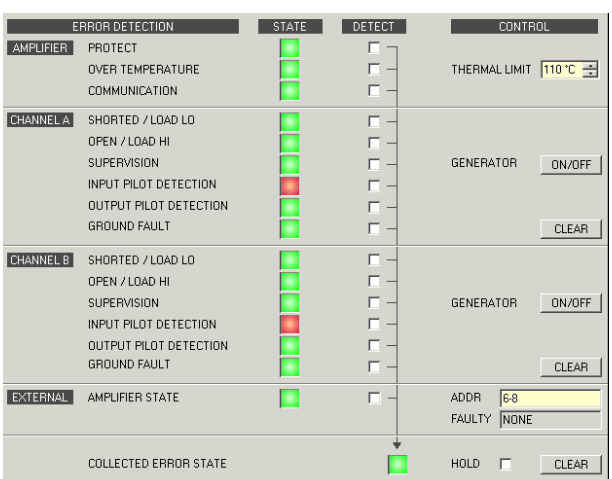
Element	Default	Range	Description
	SINE	SINE, WHITE, PINK	Type selects the test tone's signal-type. Available choices are: sine signal, white noise or pink noise.
	1000 Hz	20...20000 Hz	Freq defines the frequency of the sine signal. This parameter is not available when WHITE or PINK has been chosen as a test-tone signal.
	dBu	dBu, V	Unit selects the unit of the generator level. If the push-button is not engaged, the level is shown in dBu. When it is engaged, the output voltage is indicated in volts.

Element	Default	Range	Description
	0 dBu or 0.775 V	-60...+10 dBu or 0.001...2.451 V	These controls are for setting output level [dBu] or output voltage [V] of the corresponding amplifier outputs.
	OFF	OFF, ON	These ON/OFF push-buttons activate or deactivate test-tone signal output via corresponding amplifier channels. **CAUTION: Make sure to set a suitable output level, before activating the generator. Extreme output levels can lead to permanent damage of the connected loudspeaker systems!**

ERROR DETECTION

Error detection lists the individual STATE of fault indications. Errors collected are amp failure, channel failure, cable interruption, short-circuits, load deviation, ground fault, erroneous communication via the CAN bus as well as fault messages of other amps. A green STATE indicator signals normal operation. A red STATE indicator signals error detection.

If one of the corresponding DETECT boxes is marked, the state of that message is additionally included in the COLLECTED ERROR STATE. When activating the HOLD option, the indicator stays red after the occurrence of an error. If the HOLD option is not active, indication returns to green, once the fault is not detected anymore. Pressing the CLEAR button in the COLLECTED ERROR STATE line resets the indicator from red to green and stored errors are deleted. The COLLECTED ERROR STATE indicator resembles exactly the Amplifier State indicator of the System Check Window. The collected fault state message can be outputted via a control output. For detailed explanation please refer to chapter Config & Info.



Detailed explanation of individual fault indications

Error type	Description
PROTECT	A red Protect indicator signals that one of the amp's internal protections has been activated which usually results in auto-separating the power amps from the connected load via output relays to protect the connected loudspeaker systems and the power amp from being damaged. The Protect indicator lights for approx. 2 seconds during power-on operation.

OVER TEMPERATURE	This indicator lights red when the power amp's temperature exceeds the pre-set threshold, which defaults to 110 °C. If necessary, changing the temperature threshold is possible via THERMAL LIMIT. The power amp, however, is protected against thermal overload at all times, independent of the indication.
COMMUNICATION	This indicator shows whether communication at the CAN bus interface is normal (green) or when a problem exists (red). The power amp automatically detects whether commands from a PC or another central control unit are missing and signals the problem via the Communication Flag.
SHORTED / LOAD LO	This indicator lights red when the measured impedance value of the corresponding power amp output falls below a pre-set minimum or when it is shorted. Setting the minimum value is possible in the Load dialog.
OPEN / LOAD HI	This indicator lights red when the measured impedance value of the corresponding power amp output exceeds a pre-set maximum or when cable interruption is detected. Setting the maximum value is possible in the Load dialog.
SUPERVISION	Each amplifier channel can be monitored via internal pilot-tone signal. Pilot-tone signal (19 kHz, -10dBu) as well as detection and evaluation are active when the GENERATOR ON/OFF switch is engaged. The indicator lights green as long as the pilot-tone signal is detected with a sufficient level at the power amp output. A missing pilot-tone signal or a drop in its level below -14 dBu / 150 mV results in fault detection. Indication changes to red.
INPUT PILOT DETECTION	A remote amplifier's audio inputs support pilot-tone detection and evaluation. Using an externally generated pilot-tone signal allows the monitoring of audio cables and analog input stages. The threshold for 19 kHz pilot-tone evaluation is set to -40 dBu / 7.75 mV. The indicator lights green when an external pilot-tone signal coming from mixer, matrix, controller, etc. is detected. A missing pilot- tone signal or a drop in its level below the evaluation threshold causes the indicator to change to red. Only mark the DETECT box next to the indicator when an external pilot-tone signal actually exists and input monitoring has been configured.
OUTPUT PILOT DETECTION	This indicator is for amplifier monitoring via external pilot-tone signal. In that case, internal pilot-tone generation needs to be switched off to avoid interference between the two signals. Detection and evaluation is performed at the amplifier output. The indicator lights green when a 19 kHz pilot-tone signal with a level of at least -14 dBu / 150mV is detected. A missing pilot-tone signal or a drop in its level below -14 dBu (threshold) results in error detection. The indicator changes to red. **CAUTION: The externally fed pilot-tone signal passes through the entire signal path of the remote amplifier, i.e. the signal is influenced by filtering and x-over settings. When setting the external pilot-tone generator's level, make sure to mind possible amplification/attenuation applied by internal filters.**
GROUND FAULT	This indicator only exists for power amps with 100V transformer outputs (P900RT and P1200RT). Speaker cabling is realized in 100V technique, i.e. usually without ground reference - floating. If the indicator lights red, the evaluation circuit has detected a ground fault at the amplifier output or in the loudspeaker cabling. Once detected, a ground fault is stored in the power amp. After remedying the cause of the ground fault, press the CLEAR button to clear the indication.

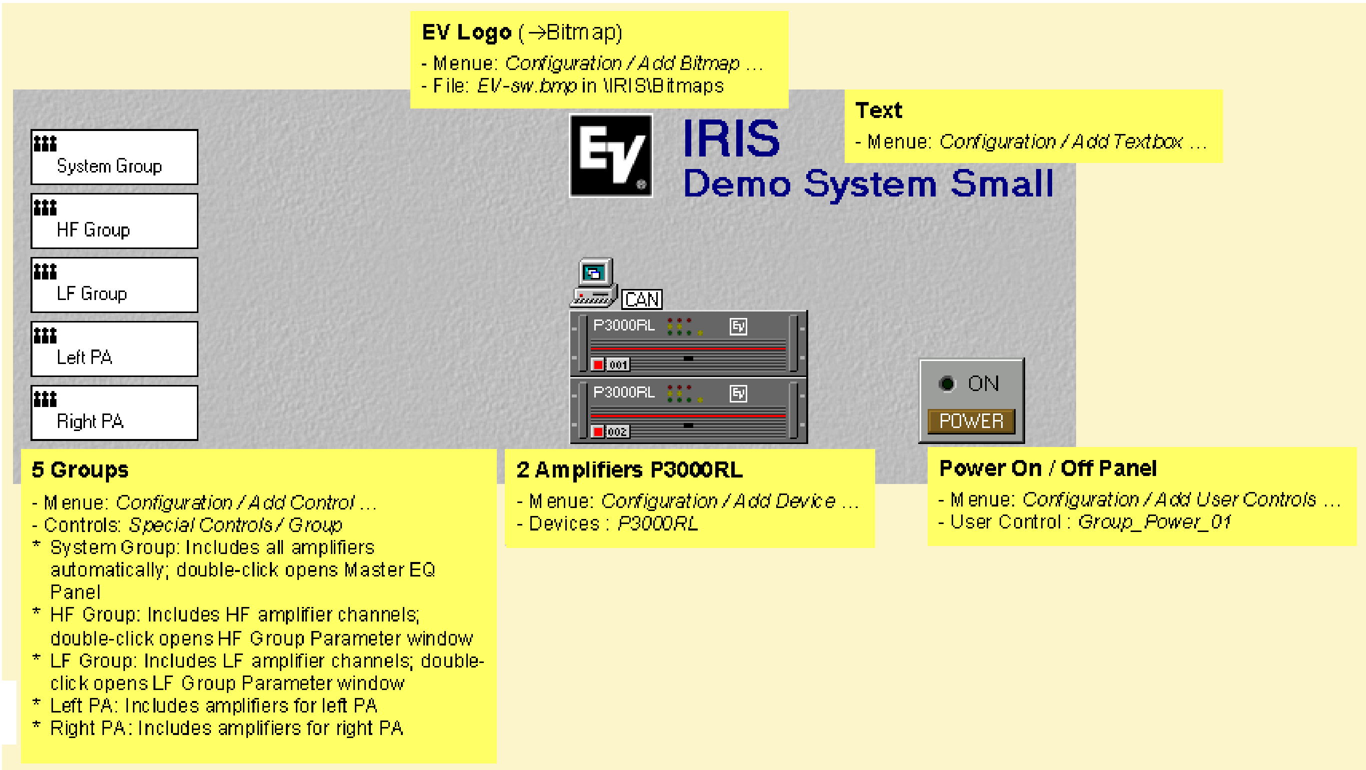
EXTERNAL AMPLIFIER STATE	A RCM-24 remote amplifier is capable of detecting and indicating the operational state of other RCM-24 amps within a CAN net- work. The addresses of all amps that are to be monitored are entered in the ADDR field, e.g. 2-4,6,11. The FAULTY field indicates the amp addresses for which errors have been detected and the COLLECTED ERROR STATE has been activated (red). The indicator changes to red as soon as at least one amplifier in the list shows erroneous operation.
COLLECTED ERROR STATE	COLLECTED ERROR STATE is a collected fault message that combines all error types detected for which the DETECT box had been marked. The HOLD function allows keeping the COLLECTED ERROR STATE for later evaluation while CLEAR clears the indication after remedying the cause of the fault. The COLLECTED ERROR STATE indication is identical to the indication in the Amplifier Status column within the System Check Window. The collected fault message can be outputted via local control output and can also be used to transmit a fault message to other power amps, where it is detected via EXTERNAL AMPLIFIER STATE.

System Examples

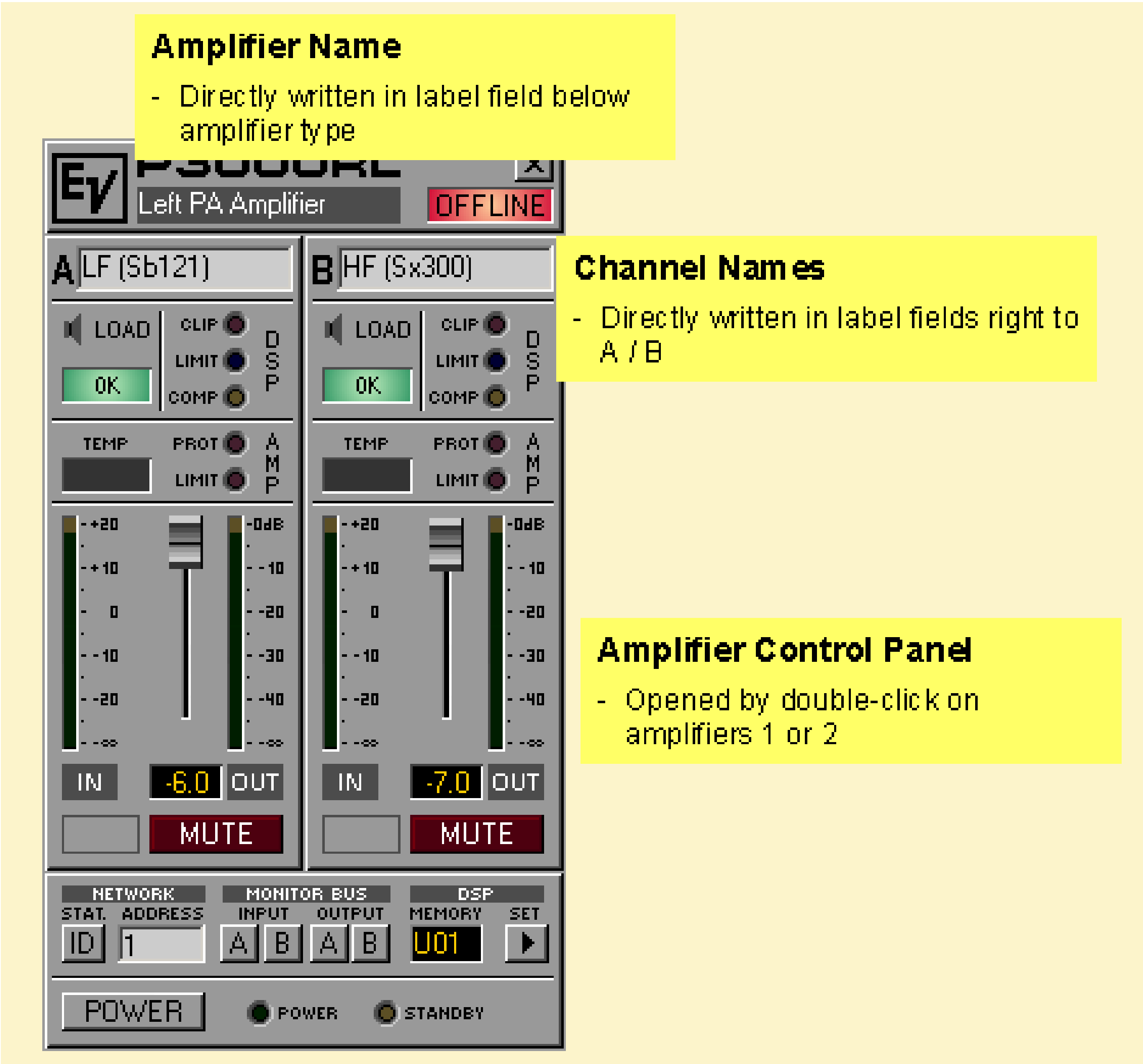
DEMO SYSTEM SMALL

The IRIS-Net project ”Demo System Small” is an example of a small EV remote amplifier system installation. This 2-way stereo system includes the following equipment: 2x P3000RL, 2x Sx300 and 2x Sb121. This example is meant to provide the necessary information to understand the entire set-up and configuration flexibility offered through the use of the IRIS-Net software for such a system. The corresponding project file ”Demo System Small.ds” is located in the following directory: ”\IRIS-Net\Projects\Demo System Small”.
The following documentation provides information of individual IRIS-Net pages, panels and parameter windows that have been created and/or used for the realization of this project. Additional explanations for the bitmap graphics are provided in yellow-shaded text fields.

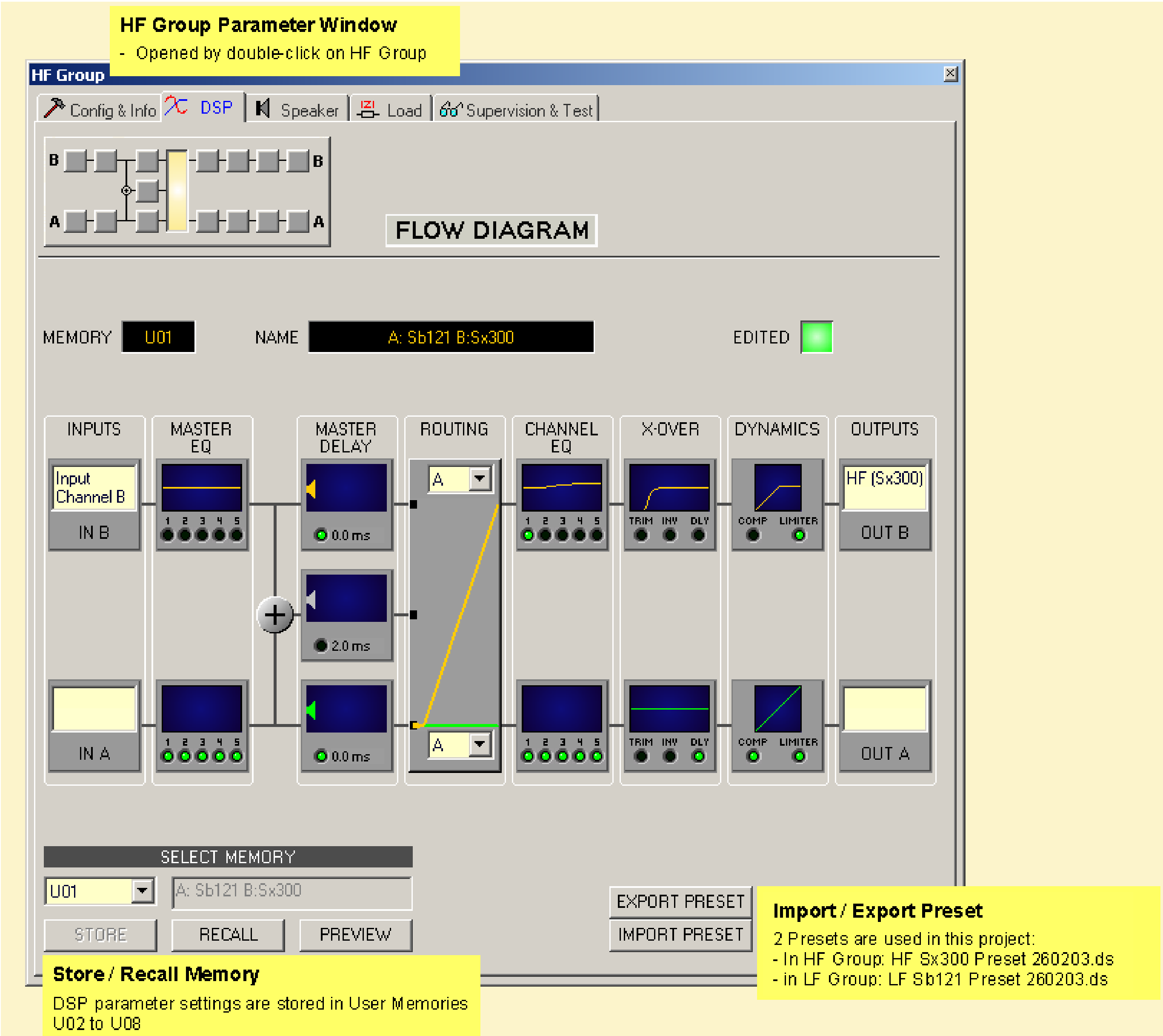
Layer 1: Configuration Page


EV Logo (→Bitmap)
- Menue: Configuration / Add Bitmap ...
- File: EV-sw.bmp in \IRIS\Bitmaps
Text
- Menue: Configuration / Add Textbox ...
IRIS
Demo System Small
System Group
HF Group
LF Group
Left PA
Right PA
CAN
P3000RL
001
P3000RL
002
ON
POWER
5 Groups
- Menue: Configuration / Add Control ...
- Controls: Special Controls / Group
* System Group: Includes all amplifiers automatically; double-click opens Master EQ Panel
* HF Group: Includes HF amplifier channels; double-click opens HF Group Parameter window
* LF Group: Includes LF amplifier channels; double-click opens LF Group Parameter window
* Left PA: Includes amplifiers for left PA
* Right PA: Includes amplifiers for right PA
2 Amplifiers P3000RL
- Menue: Configuration / Add Device ...
- Devices : P3000RL
Power On / Off Panel
- Menue: Configuration / Add User Controls ...
- User Control : Group_Power_01

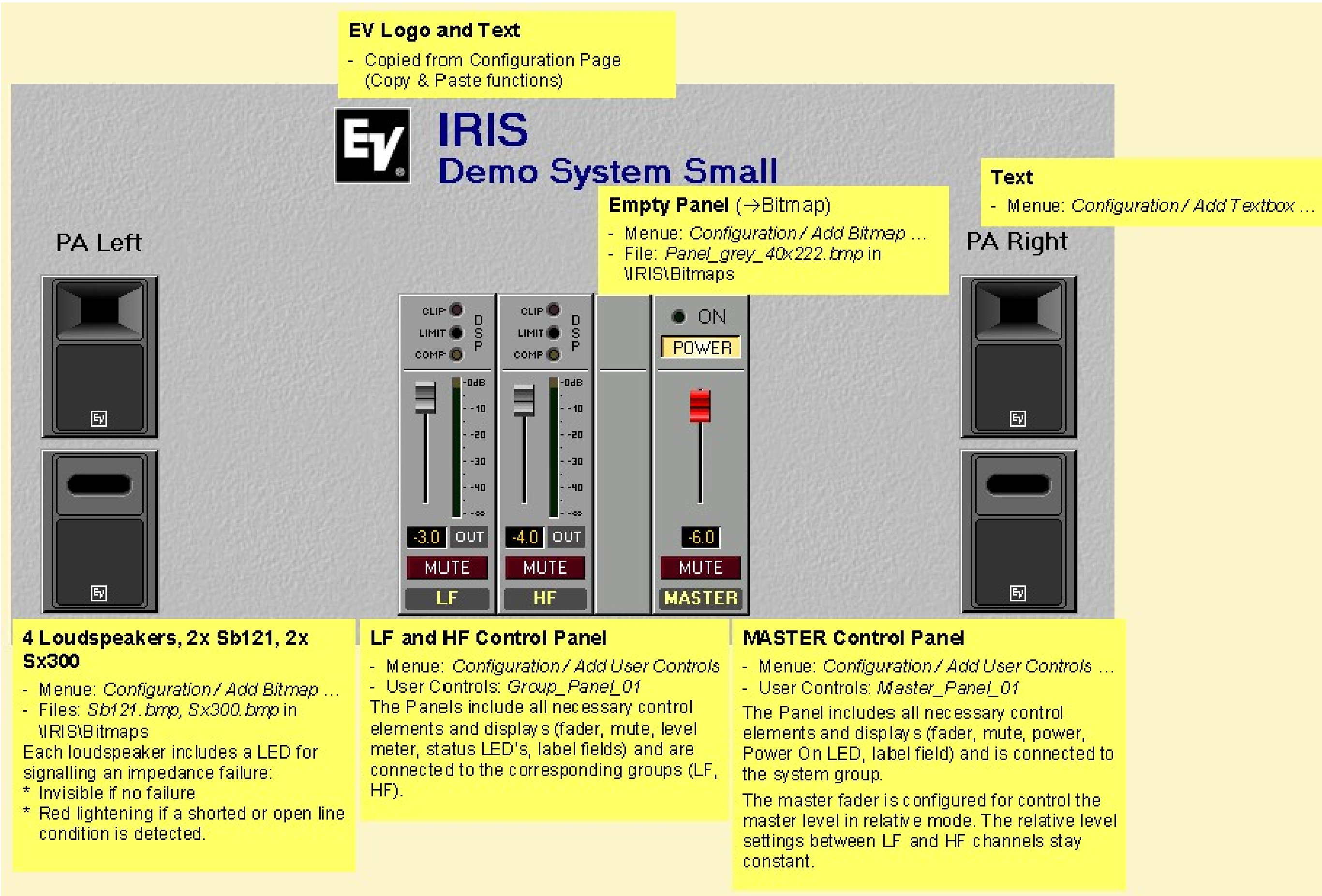
Amplifier Control Panel



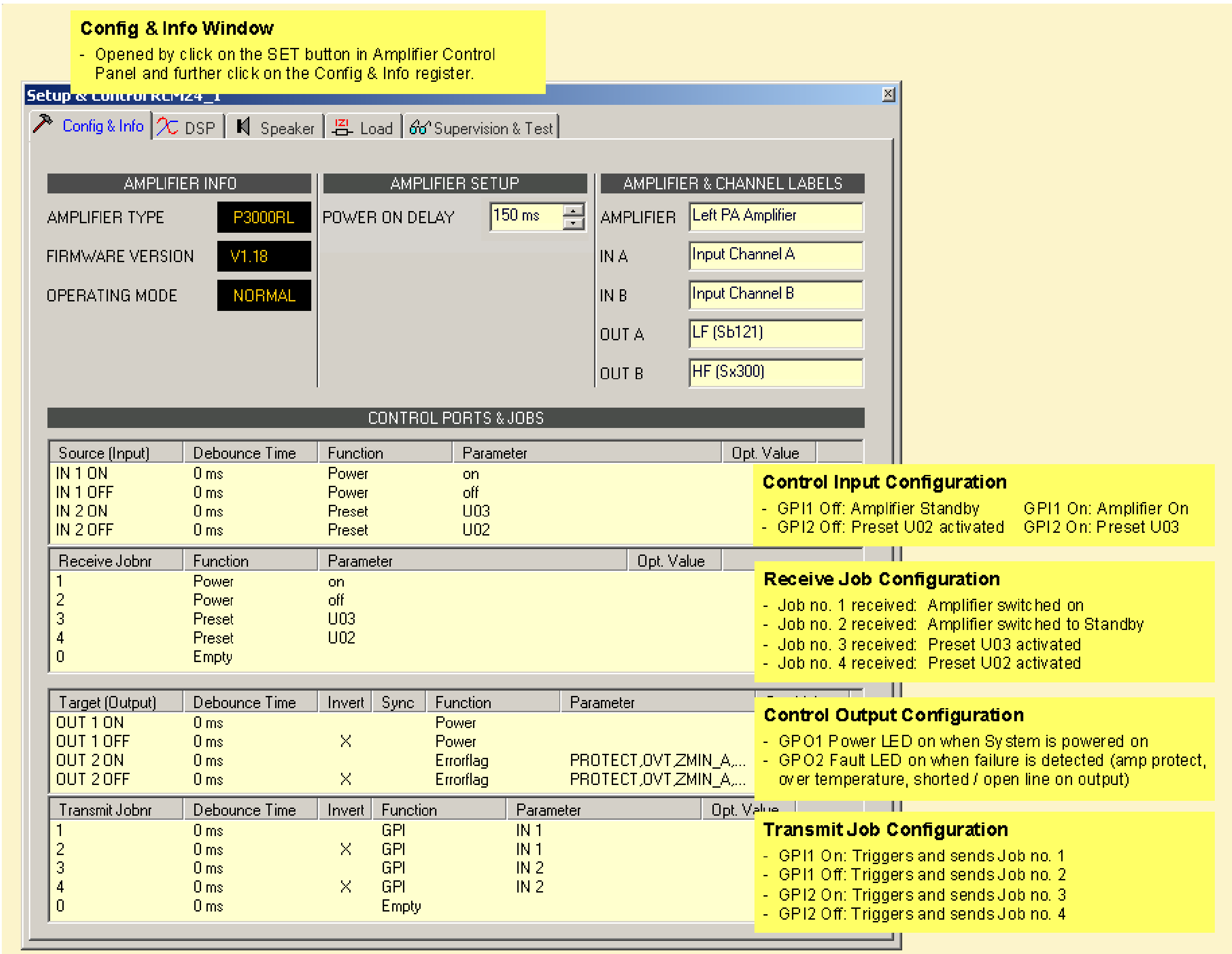
DSP Parameter Window


HF Group Parameter Window
- Opened by double-click on HF Group
HF Group
Config & Info
DSP
Speaker
Load
Supervision & Test
FLOW DIAGRAM
MEMORY
U01
NAME
A: Sb121 B:Sx300
EDITED
INPUTS
MASTER EQ
MASTER DELAY
ROUTING
CHANNEL EQ
X-OVER
DYNAMICS
OUTPUTS
Input Channel B
IN B
IN A
0.0 ms
2.0 ms
0.0 ms
TRIM
INV
DLY
COMP
LIMITER
HF (Sx300)
OUT B
OUT A
SELECT MEMORY
U01
A: Sb121 B:Sx300
STORE
RECALL
PREVIEW
EXPORT PRESET
IMPORT PRESET
Import / Export Preset
2 Presets are used in this project:
- In HF Group: HF Sx300 Preset 260203.ds
- in LF Group: LF Sb121 Preset 260203.ds
Store / Recall Memory
DSP parameter settings are stored in User Memories U02 to U08

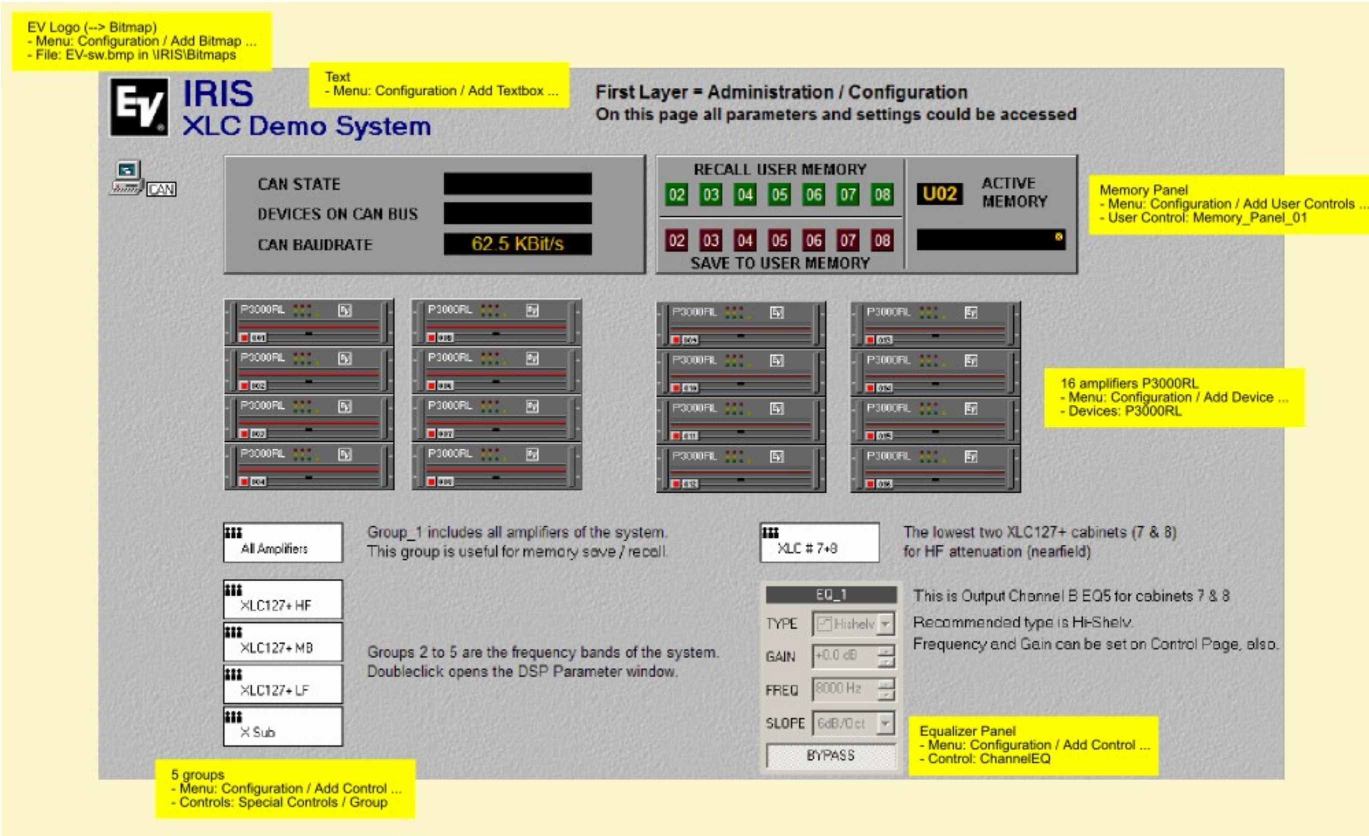
Layer 2: Control Page



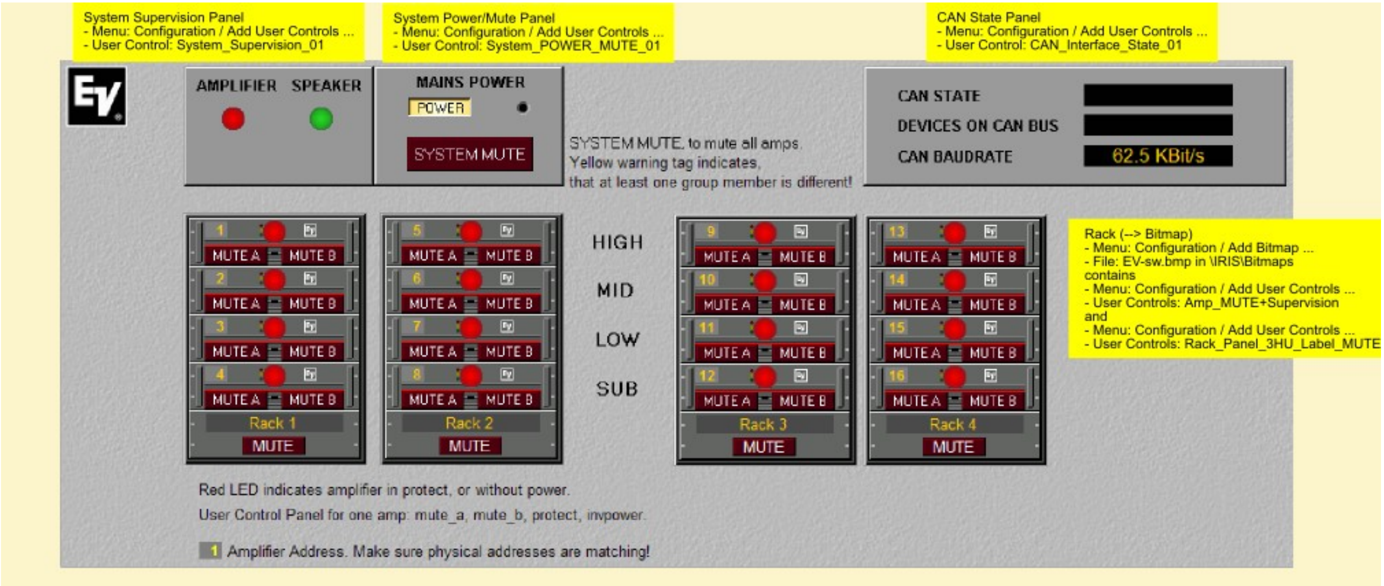
Config & Info Window



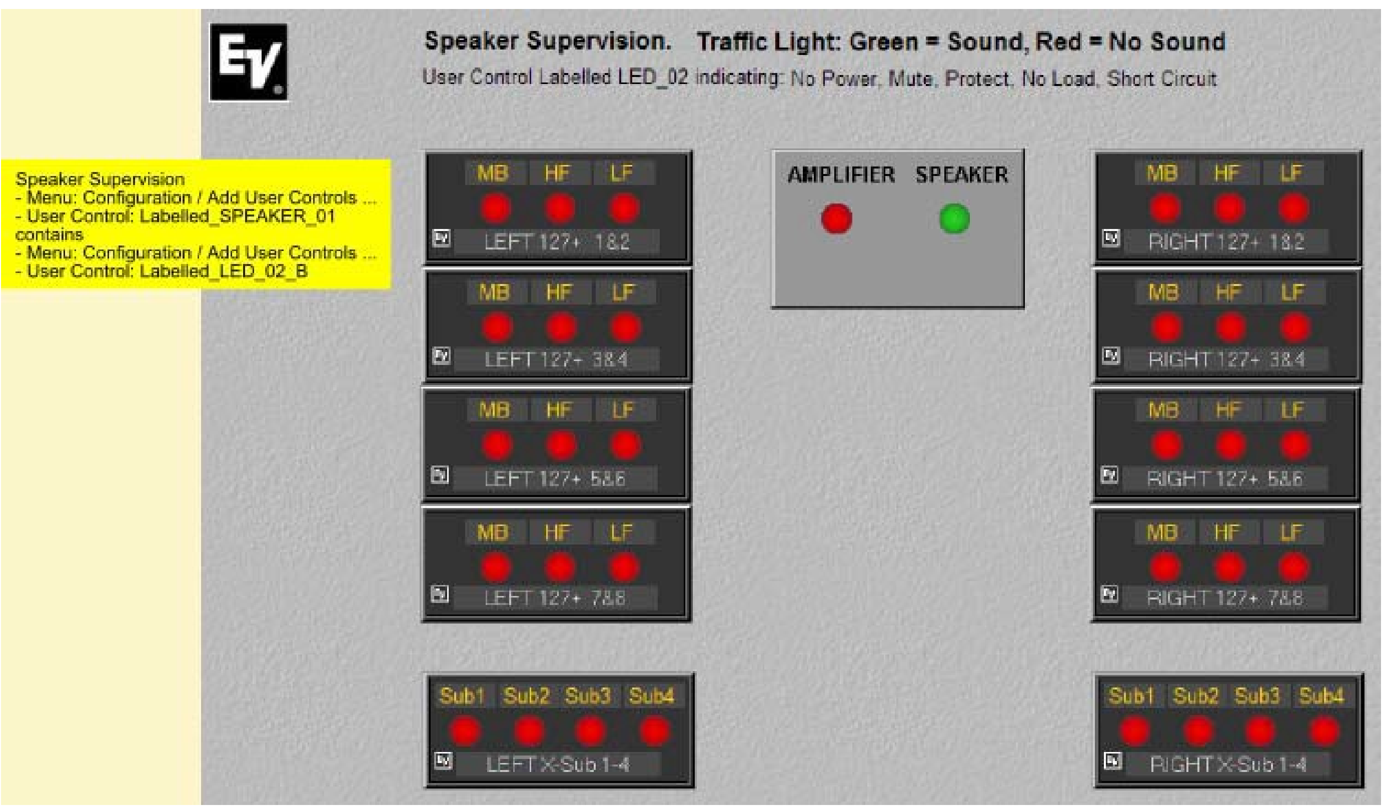
XLCDEMOPROJECT
Layer 1: Administration Page


EV Logo (--> Bitmap)
- Menu: Configuration / Add Bitmap ...
- File: EV-sw.bmp in \IRIS\Bitmaps
Text
- Menu: Configuration / Add Textbox ...
IRIS
XLC Demo System
First Layer = Administration / Configuration
On this page all parameters and settings could be accessed
CAN STATE
DEVICES ON CAN BUS
CAN BAUDRATE
62.5 KBit/s
RECALL USER MEMORY
SAVE TO USER MEMORY
U02
ACTIVE MEMORY
Memory Panel
- Menu: Configuration / Add User Controls ...
- User Control: Memory_Panel_01
16 amplifiers P3000RL
- Menu: Configuration / Add Device ...
- Devices: P3000RL
All Amplifiers
Group_1 includes all amplifiers of the system.
This group is useful for memory save / recall.
XLC # 7+8
The lowest two XLC127+ cabinets (7 & 8)
for HF attenuation (nearfield)
XLC127+ HF
XLC127+ MB
XLC127+ LF
X Sub
Groups 2 to 5 are the frequency bands of the system.
Doubleclick opens the DSP Parameter window.
EQ_1
TYPE
GAIN
FREQ
SLOPE
BYPASS
This is Output Channel B EQ5 for cabinets 7 & 8
Recommended type is Hi-Shelv.
Frequency and Gain can be set on Control Page, also.
Equalizer Panel
- Menu: Configuration / Add Control ...
- Control: ChannelEQ
5 groups
- Menu: Configuration / Add Control ...
- Controls: Special Controls / Group

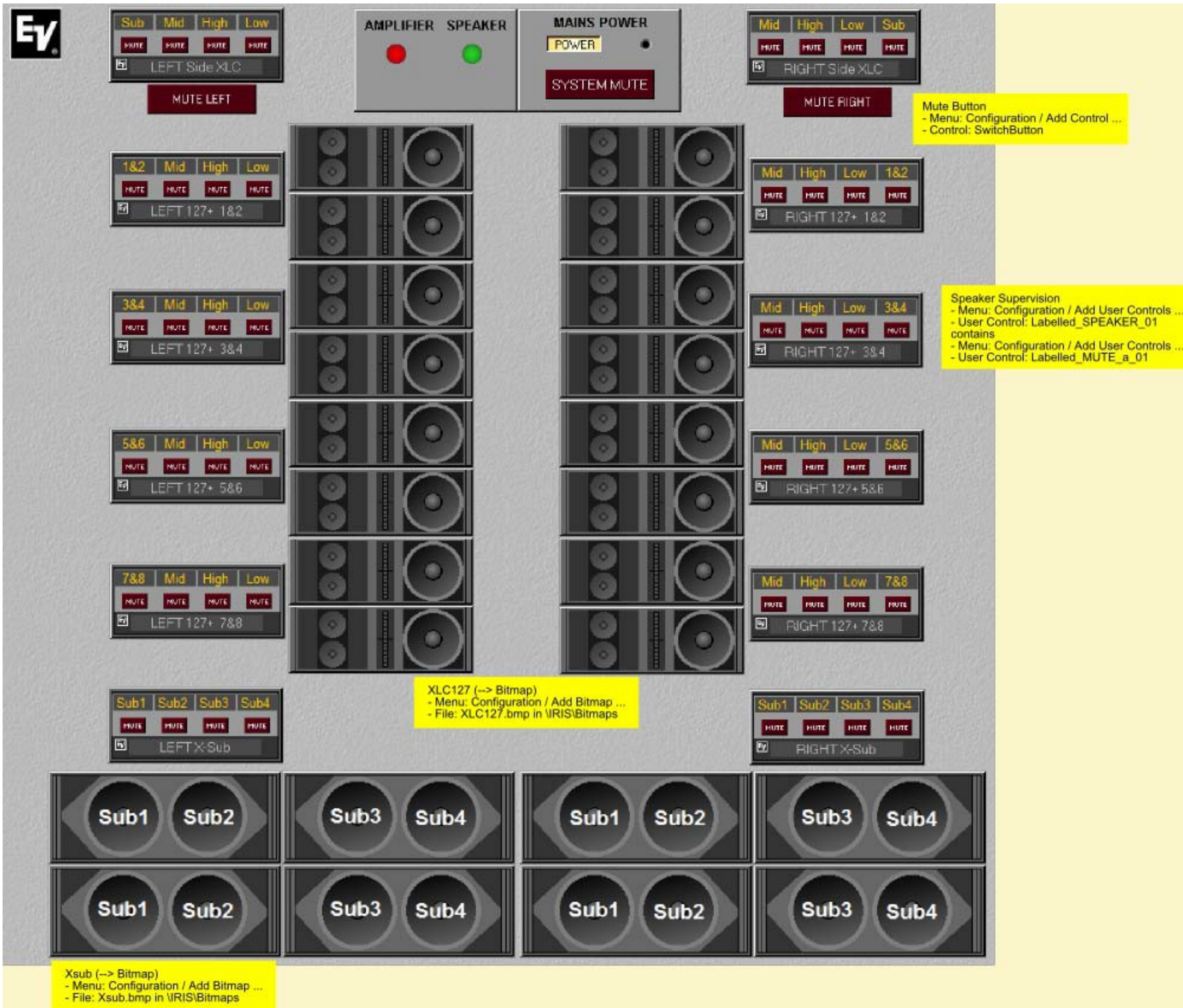
Layer 2: Amplifiers Page



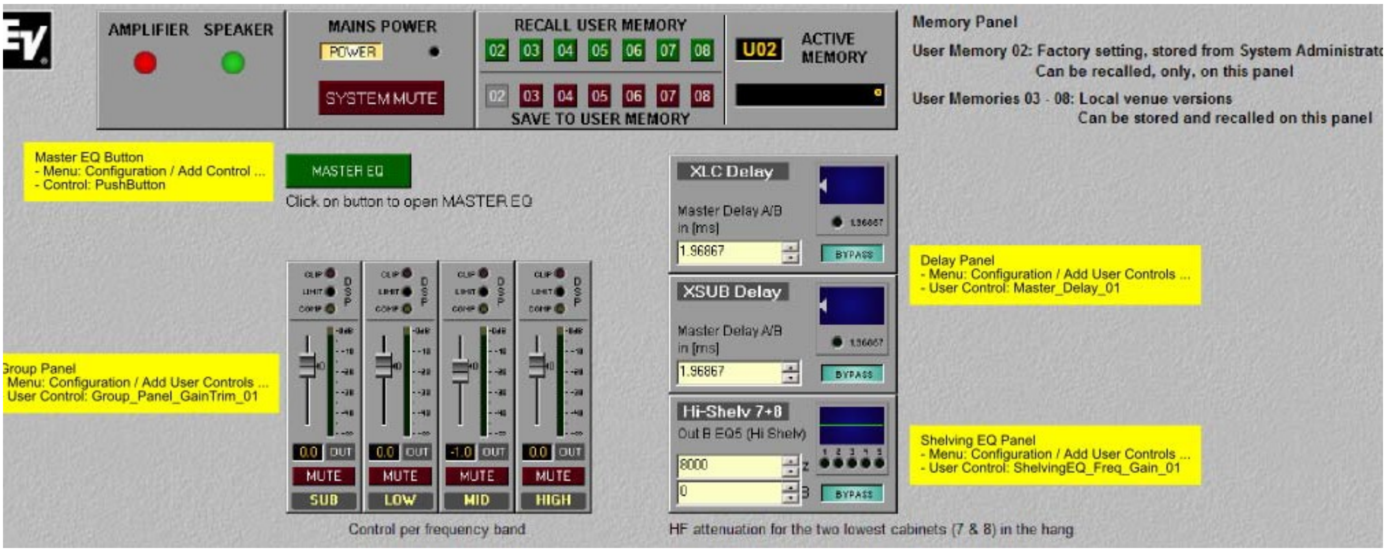
Layer 3: Speaker Supervision Page



Layer 4: Speaker Mutes Page



Layer 5: Control Page



RS-232 Protocol for EV P-Series

The RS-232 port is located on the rear panel of EV P-Series remote power amps. It can be used as interface for the connection of media control systems or facility management systems. RS-232 allows controlling and polling all parameters. Communication is performed using an easy to implement ASCII protocol which allows easy integration of remote amps in media and/or touch-panel applications. For programming notes and a complete description of the protocol please refer to the following chapters.

RS-232SETTINGS

The RS-232 interface of the RCM-24 remote power amps is permanently configured allowing full duplex operation. Set values are:

Parameter	Setting
Baud Rate	19200 bits per second
Data Bits	8
Parity	None
Stop Bits	1
Flow Control	Xon / Xoff

The command string "*** RCM-24 command mode entered ***" is sent to RS-232 after powering on the remote amp and after a short initializing period. The RS-232 interface is now ready for communication.

ASCIICONTROLPROTOCOL

A simple ASCII string protocol, which is referred to as ASCII Control Protocol is implemented in the remote amps. Commands are organized in a tree structure with up to 5 levels. The slash "/" or a space " " can be used for separation. The question mark "?" can be utilized to query parameter settings or commands of the corresponding level. To step one level down you have to enter "../".
The following table lists the ASCII Control Protocol commands with brief explanations.

Level1	Level2	Level3	Level4	Level5	Read Write	Values	
							Commands for RS232 communication
/COMM	/LINEFEED				R / W	ON, OFF	Linefeed state for RS232 communication
	/PROMPT				R / W	ON, OFF	Prompt state for RS232 communication
	/ECHO				R / W	ON, OFF	Echo state for RS232 communication
							Amplifier / Channel Names
/NAME	/AMP				R / W	up to 30 characters	Amp name
	/IN_A				R / W		Input A name
	/IN_B				R / W		Input B name
	/OUT_A				R / W		Output A name
	/OUT_B				R / W		Output B name
							Amplifier Power On / Stand-by and Operational State
/POWER	/SWITCH				R / W	ON, OFF	Switch amp ON / OFF or read out ON / OFF state
	/DELAY				R / W	0...255 [*20ms]	Power-On-Delay in steps of 20ms. 0 sets the default value, dependent on amp address.
	/STANDBY				R	ON, OFF	Read out the amp's Stand-by state
	/PROTECT				R	ON, OFF	Read out the amp's Protect state
							Connect/Disconnect Amplifier
/SERVICE	/CAN	/ CONN ECT				0...255	Transparent ASCII Control Protocol connection via CAN to remote RCM, writhe CAN address (1...250) of RCM to connect to, or write 0 to disconnect. Active remote connection is shown as address in brackets before the prompt.
							Commands for Level Indication
/METER	/AUTO				R / W	ON, OFF	Setting for automatic level meters transmission via CAN

	/QUIET				R / W	0...255 [ms]	Pause between level meters transmission via CAN
	/VU	/RLS			R / W	0...255 [ms/ dB]	Release time for Input / Output level meters
		/ INPTR S			R / W	-128...127 [dB]	Threshold for Input level meters transmission via CAN
		/ OUTT RS			R / W	-128...127 [dB]	Threshold for Output level meters transmission via CAN
	/LIMIT	/RLS			R / W	0...255 [ms/ dB]	Release time for Limiter level meters
		/TRS			R / W	-128...127 [dB]	Threshold for Limiter level meters transmission via CAN
	/READ				R		Read out all level meters
	/U				R		Read out all output voltages
	/I				R		Read out all output currents
	/Z				R / W		Read out all output impedance values including MIN / MAX values that occurred. Writing the channel name (A, B) deletes MIN / MAX values.
							Commands for Amplifier Temperature Indication
/TEMP	/ACT				R	-20...150 [°C]	Read out actual amp temperature
	/HI				R / W	20...150 [°C]	Threshold for thermal overload flag. The flag is set as soon as the temperature threshold is reached.
	/HYS				R / W	0...40 [°C]	Hysteresis for thermal overload flag. The flag is dele- ted, as soon as the temperature falls below the thres- hold minus hysteresis.
							Commands for Audio Monitoring

/MONI					R / W	NONE, RELAY, IN_A, OUT_A, IN_B, OUT_B	List of active elements for Audio Monitoring. Input and output channels can be monitored. RELAY switches active channels onto the monitor bus.
							Commands for DSP Parameters
/PRM	/IDX100				R / W		Read and write of DSP parameter values via index numbers. For further details please refer to "Descrip- tion of General DSP Parameters" and/or "DSP Parame- ter Index Table".
	...						
	/IDX1A5				R / W		
	/LOAD				R / W	1...8, F1	Load User Presets U01...U08 or Factory Presets F01. Readout of preset data loaded last. An asterisk '*' behind the preset number indicates that values have already been edited.
	/SAVE				W	1...8	Save User Preset U01...U08.
	/TITLE				R / W	bis 16 Zeichen	Preset name
	/DLYTEMP				R / W	-20.0...+60. 0 [°C]	Ambient temperature for the calculation of delays with distance values.
	/DLYUNIT				R / W	MS, SAMP- LES, FT, IN, M, CM, US, S	Delay unit. Delay values are read out with the here set unit. When writing delay values with /PRM/IDX... , the stated unit is saved together with the value.
							Commands for Control Inputs / Control Outputs
/CONTROL	/IN1	/ STATE			R	ON, OFF	State of the control input
		/ON	/TIME		R / W	0...10.0 [s]	Delay / debounce time during activation

			/FNCT		R / W	NOTHING, POWER, ABS, REL, TOG- GLE, PRESET, MONI, GFRES, MEMFLAG, MEAS, TEST- GEN	Function during activation. For further details please refer to the table "Control Inputs - GPI Functions" below.
			/PRM	/...			Parameter and values for the functions mentioned before
		/OFF	...				(same as above but for the deactivation of control inputs)
	/IN2	...					(same as above but for the control input 2)
	/OUT1	/ STATE			R / W	ON, OFF	State of the Control Output
		/ON	/TIME		R / W	0...10.0 [s]	Delay / debounce time for the programmed condition
			/FNCT		R / W	NOTHING, POWER, ABS, TEMP, VU, CTL_IN, ERR- FLAG, MEMFLAG, PRESET	Condition that activates a control output. For further details please refer to the table "Control Outputs - GPO Functions" on this page.
			/INV		R / W	ON, OFF	Inverts the result of the programmed condition
			/SYNC		R / W	ON, OFF	Lets you select whether the control outputs can be synchronized using a special CAN-command.
			/PRM	/..			Parameters and values for functions mentioned above
		/OFF	...				(same as above but for switching off a control output)
	/OUT2	...					(same as above but for the control output 2)
	/MEMFLAG	/SET			R / W	NONE, 1...16	List of currently set Memo flags

		/CLR			R / W	NONE, 1...16	List of currently reset Memo flags
							Commands for Receive and Transmit job codes
/JOB	/RX1	/ID			R / W	0...1023	Number (ID) for job code to be received. Each power amp can receive and interpret up to 5 job codes.
		/FNCT			R / W	NOTHING, POWER, ABS, REL, TOG- GLE, PRESET, MONI, GFRES, MEMFLAG, MEAS, TEST- GEN	Function when receiving a job code. For further details please refer to table "Job Codes - Receive Functions" on this page.
		/PRM	/...				Parameters and values for functions mentioned above
	...						(same as above but for receiving Job Codes 2 to 5)
	/RX5	...					
	/TX1	/ID			R / W	0...1023	Number (ID) for job code to be transmitted. Each power amp can transmit up to 5 job codes.
		/TIME			R / W	0...10.0 [s]	Delay / debounce time for programmed condition
		/FNCT			R / W	NOTHING, POWER, ABS, TEMP, VU, CTL_IN, ERR- FLAG, MEMFLAG, PRESET	Condition that triggers the transmission of a job code. For further details please refer to table "Job Codes - Transmit Functions" on this page.
		/INV			R / W	ON, OFF	Inverts the result of the programmed condition
		/PRM	/...				Parameters and values for functions mentioned above
	...						(same as above but for transmitting Job Codes 2 to 5)

	/TX5	...					
	/LAST	/RX			R / W	0000...03F F	The ID (hex code) of the last received job code is dis- played during reading. Writing simulates the reception of a job code with the stated ID (hex code) by the power amp.
		/TX			R / W	0000...03F F	The ID (hex code) of the last transmitted job code is displayed during reading. Writing transmits a job code with the stated ID (hex code).
							Commands for the Pilot Tone Generator
/PILOT	/A	/ SWITC H			R / W	ON, OFF	Pilot tone generator ON / OFF for channel A
		/ LEVEL			R / W	-128...+20 [dBu]	Pilot tone generator level for channel A
		/ ERRO R			R	ON, OFF	Channel A pilot tone error
		/ INPUT			R	ON, OFF	Pilot tone recognized at input A
		/ OUTP UT			R	ON, OFF	Pilot tone at output A
	/B	...					(same as above, but for channel B)
							Commands for Amplifier Output Load
/LOAD	/A	/MIN			R / W		Lower output load threshold for channel A (interpretation in error flag ZMIN_A)
		/MAX			R / W		Upper output load threshold for channel A (interpretation in error flag ZMAX_A)
	/B	...					(same as above, but for channel B)
	/MEAS				R W	A, B	Reading displays the actual output load values including MIN and MAX values that occurred. Writing the channel names resets MIN and MAX values.

							Commands for Error and Status Requests
/ERRFLAG	/ACT				R / W	NONE, POWER, STANDBY, PROTECT, OVT, GNDFLT_A, GNDFLT_B, ZMIN_A, ZMIN_B, ZMAX_A, ZMAX_B, PILOT_A, PILOT_B, PRE- SET, PCDUMP, DIRTY, PWR- GOOD, CAN- POLL, BRIDGED, COLLECT, GLOBAL, MEAS, Z_VLD_A, Z_VLD_B, EEPROM, PRSGATE, PLT_IN_A, PLT_OUT_A , PLT_IN_B, PLT_OUT_B	List of currently set status and error flags. Writing resets the flags GNDFLT_A, GNDFLT_B, COLL- ECT, GLOBAL, PRSGATE.
	/COLLECT				R / W		Flag template for Collected Error Flag (a list of status and error flags as mentioned above). The state is buffered (Hold function) when COLLECT is listed in the template.
	/GLBMASK				R / W	NONE, 0...255	Template for monitoring GLOBAL status and error flags of external CAN devices.
	/GLOBAL				R	NONE, 0...255	List of external CAN devices with set GLOBAL status and error flags.

							Commands for Test Generator
/SERVICE	/GEN	/A	/ SWITCH		R / W	ON, OFF	Generator ON / OFF for channel A
			/GAIN		R / W	-128...50 [dBu]	Generator output level for channel A
		/B	...		R / W		(identical parameters for channel B)
		/ MODE			R / W	SINE, WHITE, PINK	Test Generator signal type
		/ FREQ U			R / W	10.0...2000 0 .0 [Hz]	Test Generator Frequency, when SINE has been selected
		/MIX			R / W	ON, OFF	Wanted signal and Generator signal mixed (ON) or Generator solo (OFF)
		/PRE			R / W	ON, OFF	Generator signal fed in at the input or at the output

Examples:
- /POWER/SWITCH ON switches amp's power on
- /TEMP/ACT ? queries the amp temperature
- /TEMP/ACT 65 reply to query: 65 °C
- /ERRFLAG/ACT ? queries operational state and error flags
- /ERRFLAG/ACT POWER,GLOBAL reply to query: Power is On, Global Error detected (Collected Error in external CAN-devices)
- /ERRFLAG/GLOBAL ? queries for which external CAN-devices errors have been detected
- /ERRFLAG/GLOBAL 3-4 reply to query: Collected Error Flags on amps 3 and 4 are set

GENERAL DSP PARAMETER DESCRIPTION

Parameter	**Value/Settings**	**Description**
level	-128...+6 [dB]	-128dB is identical to MUTE
trim level	-30...+6 [dB]	
mute	0 / 1	0 = ON, 1 = MUTE
polarity	0 / 1	0 = Normal, 1 = Invertiert
route	0 / 1 / 2	0 = IN A, 1 = IN B, 2 = IN A+B
delay		Delay value with unit (time or distance) **HINT: Reading displays the delay value independent of the saved unit. Writing also stores the stated unit.**
bypass	0 / 1	0 = ON, 1 = BYPASS
eq type	0...5	0 = PEQ, 1 = LOSHELV, 2 = HISHELV, 3 = LOCUT, 4 = HICUT, 5 = ALLPASS

eq slope	0 / 1 / 2	0 = 0dB/Oct, 1 = 6dB/Oct, 2 = 12dB/Oct
eq frequ	20...20000 [Hz]	
eq gain	-18...+12 [dB]	
eq quality	0.4...40	EQ Quality
xover type	0...17	0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/Q0.7, 5 = 12dB/Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/ Q1.2, 8 = 12dB/Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butterworth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB-Butterworth, 17 = 24dB-Linkwitz

DSP PARAMETER INDEX TABLE

Index	Description		Index	Description
/PRM/IDX100	input A delay bypass		/PRM/IDX160	output A eq2 quality
/PRM/IDX101	input A delay		/PRM/IDX161	output A eq3 bypass
/PRM/IDX102	input A eq1 bypass		/PRM/IDX162	output A eq3 type
/PRM/IDX103	input A eq1 type		/PRM/IDX163	output A eq3 slope
/PRM/IDX104	input A eq1 slope		/PRM/IDX164	output A eq3 frequ
/PRM/IDX105	input A eq1 frequ		/PRM/IDX165	output A eq3 gain
/PRM/IDX106	input A eq1 gain		/PRM/IDX166	output A eq3 quality
/PRM/IDX107	input A eq1 quality		/PRM/IDX167	output A eq4 bypass
/PRM/IDX108	input A eq2 bypass		/PRM/IDX168	output A eq4 type
/PRM/IDX109	input A eq2 type		/PRM/IDX169	output A eq4 slope
/PRM/IDX10A	input A eq2 slope		/PRM/IDX16A	output A eq4 frequ
/PRM/IDX10B	input A eq2 frequ		/PRM/IDX16B	output A eq4 gain
/PRM/IDX10C	input A eq2 gain		/PRM/IDX16C	output A eq4 quality
/PRM/IDX10D	input A eq2 quality		/PRM/IDX16D	output A eq5 bypass
/PRM/IDX10E	input A eq3 bypass		/PRM/IDX16E	output A eq5 type
/PRM/IDX10F	input A eq3 type		/PRM/IDX16F	output A eq5 slope
/PRM/IDX110	input A eq3 slope		/PRM/IDX170	output A eq5 frequency
/PRM/IDX111	input A eq3 frequ		/PRM/IDX171	output A eq5 gain
/PRM/IDX112	input A eq3 gain		/PRM/IDX172	output A eq5 quality
/PRM/IDX113	input A eq3 quality		/PRM/IDX173	output B level
/PRM/IDX114	input A eq4 bypass		/PRM/IDX174	output B trim level
/PRM/IDX115	input A eq4 type		/PRM/IDX175	output B delay bypass

/PRM/IDX116	input A eq4 slope		/PRM/IDX176	output B delay
/PRM/IDX117	input A eq4 frequ		/PRM/IDX177	output B mute
/PRM/IDX118	input A eq4 gain		/PRM/IDX178	output B polarity
/PRM/IDX119	input A eq4 quality		/PRM/IDX179	output B route
/PRM/IDX11A	input A eq5 bypass		/PRM/IDX17A	output B compressor bypass
/PRM/IDX11B	input A eq5 type		/PRM/IDX17B	output B compressor type (ratio)
/PRM/IDX11C	input A eq5 slope		/PRM/IDX17C	output B compressor threshold
/PRM/IDX11D	input A eq5 frequ		/PRM/IDX17D	output B compressor attack
/PRM/IDX11E	input A eq5 gain		/PRM/IDX17E	output B compressor release
/PRM/IDX11F	input A eq5 quality		/PRM/IDX17F	output B limiter bypass
/PRM/IDX120	input B delay bypass		/PRM/IDX180	output B limiter threshold
/PRM/IDX121	input B delay		/PRM/IDX181	output B limiter release
/PRM/IDX122	input B eq1 bypass		/PRM/IDX182	output B xover hipass type
/PRM/IDX123	input B eq1 type		/PRM/IDX183	output B xover hipass frequ
/PRM/IDX124	input B eq1 slope		/PRM/IDX184	output B xover lopass type
/PRM/IDX125	input B eq1 frequ		/PRM/IDX185	output B xover lopass frequ
/PRM/IDX126	input B eq1 gain		/PRM/IDX186	output B eq1 bypass
/PRM/IDX127	input B eq1 quality		/PRM/IDX187	output B eq1 type
/PRM/IDX128	input B eq2 bypass		/PRM/IDX188	output B eq1 slope
/PRM/IDX129	input B eq2 type		/PRM/IDX189	output B eq1 frequ
/PRM/IDX12A	input B eq2 slope		/PRM/IDX18A	output B eq1 gain
/PRM/IDX12B	input B eq2 frequ		/PRM/IDX18B	output B eq1 quality
/PRM/IDX12C	input B eq2 gain		/PRM/IDX18C	output B eq2 bypass
/PRM/IDX12D	input B eq2 quality		/PRM/IDX18D	output B eq2 type
/PRM/IDX12E	input B eq3 bypass		/PRM/IDX18E	output B eq2 slope
/PRM/IDX12F	input B eq3 type		/PRM/IDX18F	output B eq2 frequ
/PRM/IDX130	input B eq3 slope		/PRM/IDX190	output B eq2 gain

/PRM/IDX131	input B eq3 frequ		/PRM/IDX191	output B eq2 quality
/PRM/IDX132	input B eq3 gain		/PRM/IDX192	output B eq3 bypass
/PRM/IDX133	input B eq3 quality		/PRM/IDX193	output B eq3 type
/PRM/IDX134	input B eq4 bypass		/PRM/IDX194	output B eq3 slope
/PRM/IDX135	input B eq4 type		/PRM/IDX195	output B eq3 frequ
/PRM/IDX136	input B eq4 slope		/PRM/IDX196	output B eq3 gain
/PRM/IDX137	input B eq4 frequ		/PRM/IDX197	output B eq3 quality
/PRM/IDX138	input B eq4 gain		/PRM/IDX198	output B eq4 bypass
/PRM/IDX139	input B eq4 quality		/PRM/IDX199	output B eq4 type
/PRM/IDX13A	input B eq5 bypass		/PRM/IDX19A	output B eq4 slope
/PRM/IDX13B	input B eq5 type		/PRM/IDX19B	output B eq4 frequ
/PRM/IDX13C	input B eq5 slope		/PRM/IDX19C	output B eq4 gain
/PRM/IDX13D	input B eq5 frequ		/PRM/IDX19D	output B eq4 quality
/PRM/IDX13E	input B eq5 gain		/PRM/IDX19E	output B eq5 bypass
/PRM/IDX13F	input B eq5 quality		/PRM/IDX19F	output B eq5 type
/PRM/IDX140	input A+B delay bypass		/PRM/IDX1A0	output B eq5 slope
/PRM/IDX141	input A+B delay		/PRM/IDX1A1	output B eq5 frequ
/PRM/IDX142	output A level		/PRM/IDX1A2	output B eq5 gain
/PRM/IDX143	output A trim level		/PRM/IDX1A3	output B eq5 quality
/PRM/IDX144	output A delay bypass		/PRM/IDX1A4	output A configuration
/PRM/IDX145	output A delay		/PRM/IDX1A5	output B configuration
/PRM/IDX146	output A mute			
/PRM/IDX147	output A polarity			
/PRM/IDX148	output A route			
/PRM/IDX149	output A compressor bypass			
/PRM/IDX14A	output A compressor type (ratio)			
/PRM/IDX14B	output A compressor threshold			
/PRM/IDX14C	output A compressor attack			
/PRM/IDX14D	output A compressor release			

/PRM/IDX14E	output A limiter bypass			
/PRM/IDX14F	output A limiter threshold			
/PRM/IDX150	output A limiter release			
/PRM/IDX151	output A xover hipass type			
/PRM/IDX152	output A xover hipass frequ			
/PRM/IDX153	output A xover lopass type			
/PRM/IDX154	output A xover lopass frequ			
/PRM/IDX155	output A eq1 bypass			
/PRM/IDX156	output A eq1 type			
/PRM/IDX157	output A eq1 slope			
/PRM/IDX158	output A eq1 frequ			
/PRM/IDX159	output A eq1 gain			
/PRM/IDX15A	output A eq1 quality			
/PRM/IDX15B	output A eq2 bypass			
/PRM/IDX15C	output A eq2 type			
/PRM/IDX15D	output A eq2 slope			
/PRM/IDX15E	output A eq2 frequ			
/PRM/IDX15F	output A eq2 gain			

Control Inputs - GPI Functions

Every control input can be programmed with individual functions for switching on (/CONTROL/INx/ON/...) and switching off (/CONTROL/INx/OFF/...). When the state of a control input changes, the programmed function is executed after the previously set delay or debounce times are expired (up to 10 sec.). Available functions are explained in the following table.

Job Codes - Receive Functions

Job codes are distributed throughout the CAN network via broadcast commands. Each job code has a freely definable number (ID). Received job codes can trigger the same functions as local GPI control inputs. Receiving a job code with the defined number (ID) triggers the function with its specified parameter values. Available functions for /JOB/RXx/FNCT/... and corresponding parameters /JOB/RXx/PRM/... are identical with local GPI functions, as outlined in the table.

Function	Parameter	Range	Description
NOTHING			No function
POWER			Controls Power On / Stand-by
	../PRM/ SWITCH	ON	Switched the amp's power to ON
		OFF	Switches the amp in Stand-by mode
		FLIP	Toggles between ON and Stand-by and vice versa
ABS			Sets the selected DSP parameter to an absolute value
	../PRM/IDX	100...1A5	Selects the DSP parameter via index number
	../PRM/ VALUE		Relative change of the parameter
REL			Changes the selected DSP parameter in relation to the actual value
	../PRM/IDX	100...1A5	Selects the DSP parameter via index number
	../PRM/ VALUE		Relative change of the parameter
TOGGLE			Toggles a DSP parameter between 0 and 1 (this only makes sense for flag parameters, e.g. MUTE, BYPASS, etc.)
	../PRM/IDX	100...1A5	Selects the DSP parameter via index number
PRESET			Loads a DSP preset
	../PRM/NR	1...8, F1	Selects an user preset U01 to U08 or a factory preset F01
MONI			Controls the selection for the audio monitoring bus
	../PRM/SEL	NONE, RELAY, IN_A, OUT_A, IN_B, OUT_B	Selects audio monitoring parameters. All combinations are possible.
	../PRM/ SWITCH	ON, OFF	Switches the selected audio monitoring parameter ON or OFF
GFRES			Deletes stored Ground Fault flags in selected channels
	../PRM/CHAN	A, B	Any combination of output channels is possible
MEMFLAG			Manipulates general Memo flags
	../PRM/CLR	NONE, 1...16	Clears selected flags
	../PRM/ TOGGLE	NONE, 1...16	Changes the state of selected flags. Use CLR and TOGGLE together, so that selected flags are set afterwards.
MEAS			Initiates impedance testing at a fixed frequency
	../PRM/ FREQU	10...20000 [Hz]	Generator frequency for impedance test
	../PRM/ GAIN_A	-128...+50 [dBu]	Generator level for impedance test in channel A

	../PRM/ GAIN_B	-128...+50 [dBu]	Generator level for impedance test in channel B
	../PRM/TIME	0.0, 0.1...4.17 [ms]	Impedance test time span. 0.0 = continuously ON
	../PRM/MIX	ON, OFF	Wanted signal and Generator signal mixed
	../PRM/PRE	ON, OFF	Generator signal fed in at the input (On) or output (Off) of the DSP signal chain
TESTGEN			Defines parameters for the audio testing generator
	../PRM/A/ SWITCH	ON, OFF	Switches the testing generator at channel A ON
	../PRM/A/ GAIN	-128...+50 [dBu]	Defines the testing generator output level for channel A
	../PRM/B/...		(same as above but for channel B)
	../PRM/ MODE	SINE, WHITE, PINK	Defines the testing generator's signal type
	../PRM/ FREQU	10...20000 [Hz]	Defines the generator frequency, when SINE is selected
	../PRM/MIX	ON, OFF	Wanted signal and Testing generator signal mixed
	../PRM/PRE	ON, OFF	Testing generator signal fed in at the input (On) or output (Off) of the DSP signal chain
ROUTING			Manipulates the routing parameters in both channels simultaneously
	../PRM/A	A, B, A+B, NO_CHANGE	Sets an input as audio source for output A
	../PRM/B	A, B, A+B, NO_CHANGE	Sets an input as audio source for output B

CONTROL OUTPUTS - GPO FUNCTIONS
Two conditions can be programmed for each control output which either activate the output (/CONTROL/OUTx/ON/...) or deactivate the output (/CONTROL/OUTx/OFF/...). When the assigned function (/CONTROL/OUTx/ON/FNCT or /CONTROL/OUTx/OFF/FNCT) is recognized as "true" and the state is maintained for at least the set delay or debounce times (up to 10 sec.), the control output changes to activated (On) or deactivated (Off). The INV parameter allows inverting the state of the assigned function. Synchronizing the switching of control outputs is possible by means of a special system-wide CAN command, when SYNC is set to ON. Available functions and corresponding settings are explained in the following table.

JOB CODES - TRANSMIT FUNCTIONS
Job codes are distributed throughout the CAN network via broadcast command. Each job code has a freely definable number (ID). Identical conditions can be assigned to job codes and control outputs. A job code with a defined number (ID) is transmitted, when the corresponding condition for (/JOB/TXx/FNCT) is recognized as "true" and the state is maintained for at least the set delay or debounce times (up to 10 sec.). The INV parameter allows inverting the state of the assigned function. Available functions for /JOB/TXx/FNCT/... as well as corresponding parameters /JOB/TXx/PRM/... are identical to local GPO functions, as outlined in the table.

Function	Parameter	Range	Description
NOTHING			No function
POWER			Interpretation results in "true", when the power amp is powered on (even during power-on delay) and "false", when the amp's power is off.
ABS			Interpretation results in "true", when the DSP parameter value is higher or equals the reference value.
	../PRM/IDX	100...1A5	Selects the DSP parameter via index number
	../PRM/ VALUE		Reference value
TEMP			Interpretation results in "true", when the measured amplifier temperature is higher or equals the reference value.
	../PRM/ CELSIUS	-20...150 [°C]	Temperature reference value
VU			Interpretation results in "true", when at least one of the selected values is higher or equals the programmed reference value.
	../PRM/SEL	IN_A, OUT_A, ALIM_A, DLIM_A, COMP_A, IN_B. OUT_B, ALIM_B, DLIM_B, COMP_B	Any combination of the values listed is possible. ALIM = Amplifier Limiter DLIM = DSP Limiter COMP = DSP Compresser
	../PRM/DB	[dB]	VU reference value
CTL_IN			Interpretation results in "true", when the selected control input is activated.
	../PRM/IDX	1, 2	Selects a control input
ERRFLAG			Interpretation results in "true", when one of the selected flags is set. Any combination of the flags listed is possible.
	../PRM/MASK	POWER	set, when the power is OFF
		STANDBY	set, when the amplifier is in Stand-by mode
		PROTECT	set, when the amp's Protect mode is activated
		OVT	set, when the amp's thermal limit is exceeded
		GNDFLT_A, GNDFLT_B	set, when ground fault has been detected
		ZMIN_A, ZMIN_B, ZMAX_A, ZMAX_B	set, when the measured output load is out of limit value range

		PILOT_A, PILOT_B	set, when pilot tone monitoring returns errors
		DIRTY	set, when the actual preset has been edited but has not been saved yet
		PWRGOOD	set, when Power Good interpretation returns errors
		CANPOLL	set, when the CAN Polling timed out
		BRIDGED	set, when the power amp is operated in Bridged Mode (only with P3000RL)
		COLLECT	set, when the amp's Collected Error Flag is ON
		GLOBAL	set, when the amp's External Amplifier Error Flag is ON
		MEAS	set, when the internal testing generator has been activated for output load measurement
		Z_VLD_A, Z_VLD_B	set, when output load measuring is not possible because of missing or too low signal
		EEPROM	set when there is an error in EEPROM administration
		PRSGATE	if not set, only limited preset changes are possible
		PLT_IN_A, PLT_IN_B	set, when the 19kHz pilot tone signal applied to the amplifier input is not recognized
		PLT_OUT_A, PLT_OUT_B	set, when the 19kHz pilot tone signal applied to the amplifier output is not recognized
MEMFLAG			Interpretation results in "true", when the actual state of the selected memo flags resembles the reference pattern.
	../PRM/MASK	NONE, 1...16	Selects memo flags to be interpreted (listing)
	../PRM/VALUE	NONE, 1...16	Defines the expected reference pattern for memo flags
PRESET			Interpretation results in "true", when the actual preset is identical to a selected preset.
	../PRM/DIRTY	ON, OFF	Selection is also valid, when parameters have been changed (dirty)
	../PRM/USER	NONE, 1...8	List of selected user presets
	../PRM/FACT	NONE, 1	List of selected factory presets

Firmware Upgrade

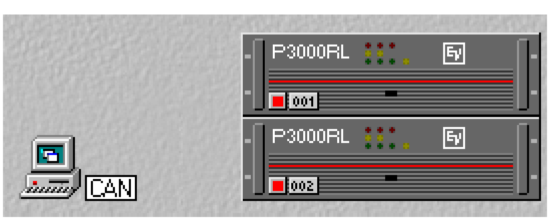
The firmware of EV remote amps is stored in a FLASH-memory chip. This technology has been chosen to be able to provide the users with new software without the hassle of physically exchanging memory chips inside of a remote amplifier. Using IRIS-Net, upgrading the firmware is possible via the CAN Remote Control Interface. In this way you can install new firmware and future software extensions to always keep your EV Remote Amplifier System up-to-date.

Caution!
Upgrading the firmware is always a very sensible procedure – comparable to updating the OS in the FLASH-memory of a PC. Therefore, obeying the following precautions and instructions is absolutely mandatory:
Consequences

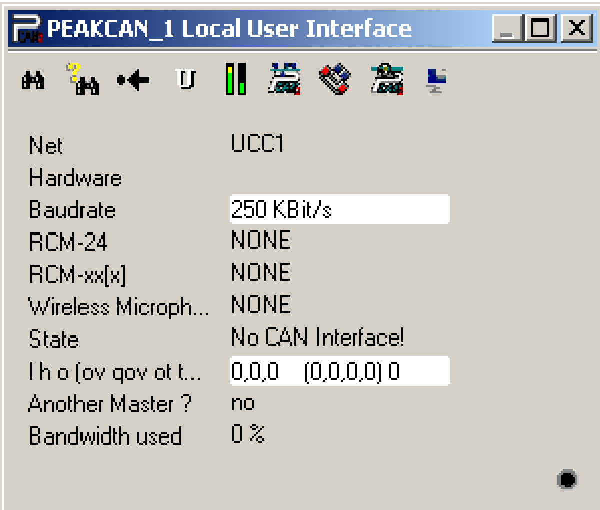
1. Make sure your mains supply is absolutely stable and the mains fuse is sufficiently dimensioned to be capable of handling all amplifiers connected.
   Loss of power during the upgrade would result in the firmware installation being incomplete or deleted and the remote amplifier could not be operated. In such a case installing the firmware is only possible through the use of a special FLASH boot loader via RS-232 interface.
   For details please contact an authorized service center or our technical support.
2. Simultaneously upgrading the firmware of more than four remote amplifiers is not recommended. Performing a firmware-upgrade for the first time, only connect a single remote amp. Once you are familiar with the upgrading procedure, you can connect and update 2, 3 or 4 amps at the same time.
3. Only connect the remote amps to the CAN Remote Control network that are to be updated. Disconnect any other remote amps from the CAN-bus during the upgrade. Make sure to carefully mind all regulations for the CAN Remote Control network, especially the 120 Ω termination at both ends of the bus.
4. Check the status of the CAN-interface. The parameter “State” has to show “OK”. The values of the error flags “I h o (ov ot to) wr” may not change or rise to guarantee proper remote amp connection.
5. Even when fault messages appear during the upgrade procedure, the irrevocable rule is: NEVER SWITCH OFF THE POWER OF AN AMPLIFIER THAT IS TO BE UPGRADED!
   If any fault messages are indicated, repeating the upgrade procedure for the affected remote amplifier step-by-step is possible. If in doubt or in need of assistance, please contact an service center or our technical support.

HOW TO UPGRADE THE FIRMWARE

1. Connect the desired remote amp(s) via CAN-bus to your PC.
2. Start the IRIS-Net software and open your project. Your remote amps and the icon of a PC with CAN-label should appear on your screen. The PC-icon represents the CAN-interface of your PC or notebook.



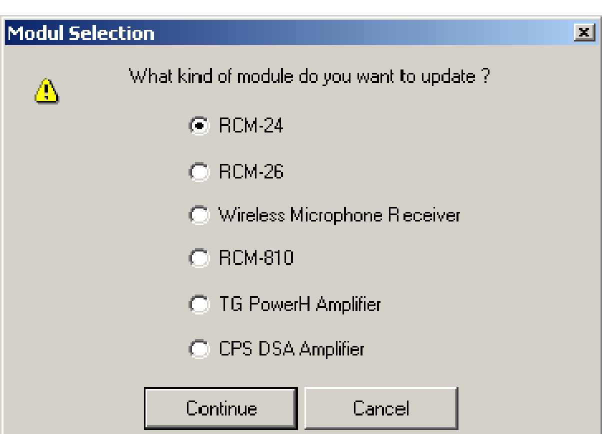
3. Double-clicking onto the PC-icon opens the CAN-interface window. CAN-bus status and connected remote amps are displayed. This window display is available in off-line mode.



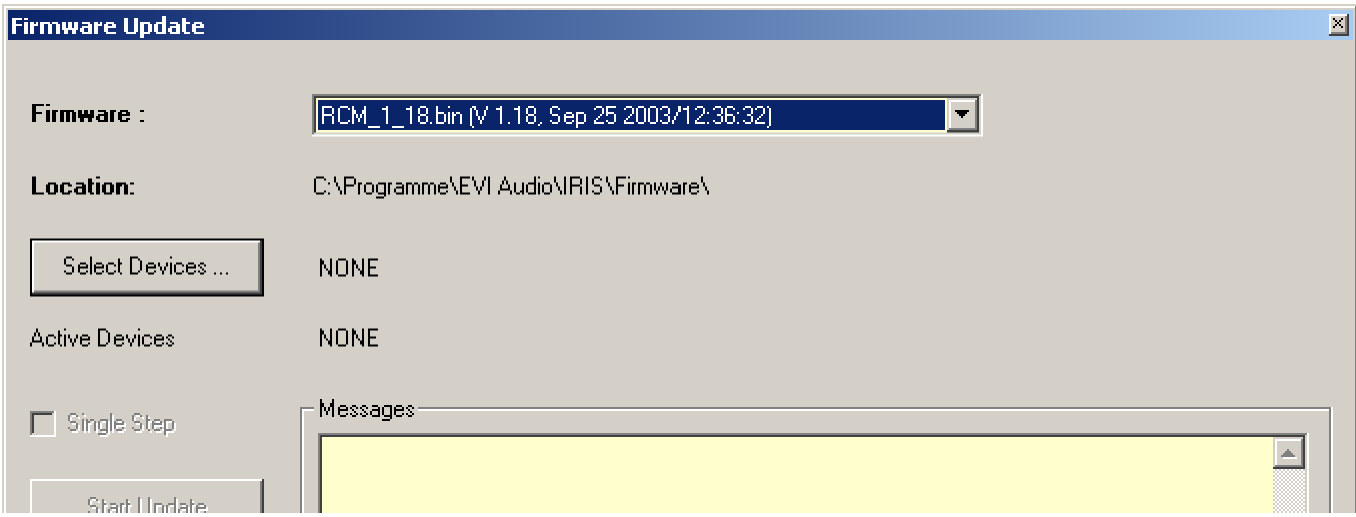
4. Make sure to check the following parameters before upgrading:

Element	Description
Baud rate	Indicates the set baud rate. Normally you don’t have to change the system’s baud rate for upgrading.
RCM-24	Indicates the addresses of the remote amps connected. Make sure that the addresses shown are only the ones of remote amps that you want to upgrade.
State	Indicates the CAN-interface status. This has to read “OK”. Otherwise, starting the firmware upgrade is not permissible.
I h o (...	Indicates different error flags. Under no circumstance the first 3 digits may rise. Clicking into the white field and entering “0” resets the error flags.
Band width used	Indicates the used bandwidth of the CAN-bus in percent. Make sure to check that the CAN-bus is not too busy, i.e. high data traffic.

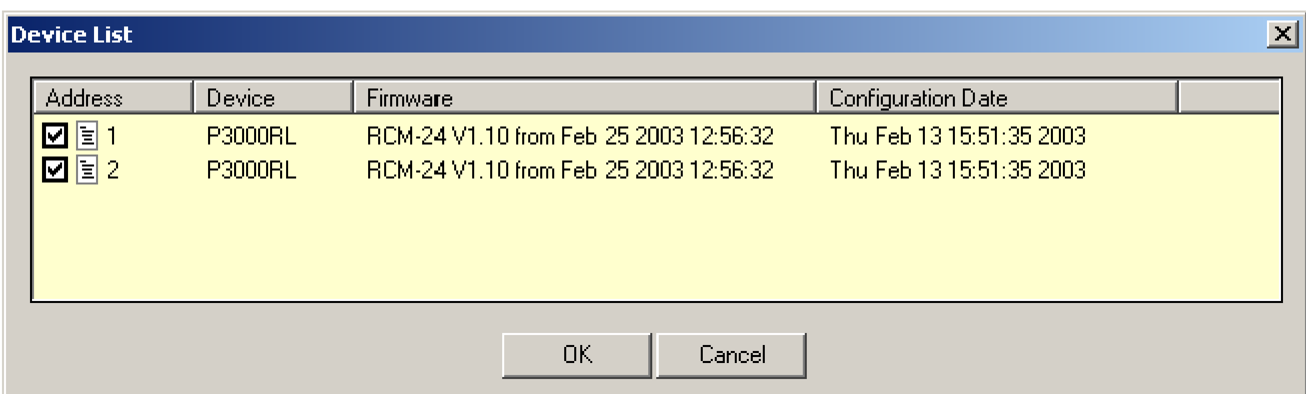
5. The CAN-interface window provides a toolbar (top line). Clicking onto the U-icon (Update) opens the Module Selection dialog. Select RCM-24 and click the Continue button.



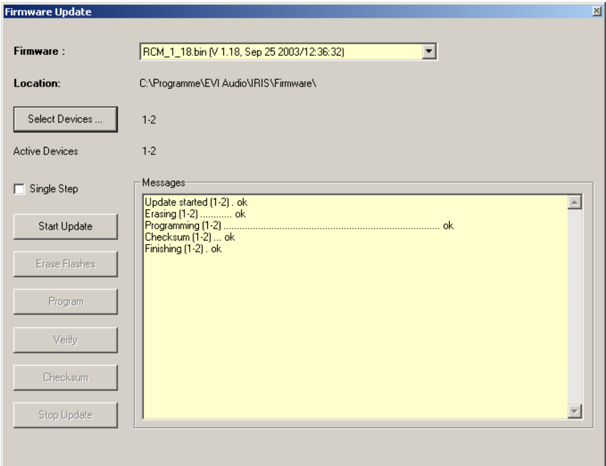
6. The actual firmware file including version number and date is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date remote amplifier firmware version. The corresponding file is located in the directory: \IRIS-Net\Firmware\RCM-24. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.



7. Click onto the button “Select Devices...” to open a list of all remote amps connected. Select the amp(s) that you want to update and click into the “OK” filed. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus. When performing the firmware upgrade for the first time, connecting only a single amplifier is recommended to become familiar with the upgrading procedure.



8. The addresses of the selected remote amp(s) are shown in the firmware update window on the right side next to the button “Select Devices...” and in the line “Active Devices”. Clicking onto “Start Update” starts the upgrade procedure. The single steps of the update are shown in the “Messages” window. The progress of some parts of the upgrade which take a little longer is indicated through dots behind the corresponding name. The message “ok” has to appear at the end of each line. The following example shows how to upgrade the firmware of the remote amps with the addresses 1 and 2 to firmware version V 1.18.



9. The message “Finishing ... ok” indicates that upgrading has been successful. The remote amp(s) are reset. Afterwards they are again ready for operation. The upgrade procedure is finished and you can close the dialog window or proceed with upgrading other remote amps.

ADDITIONAL NOTES CONCERNING A FIRMWARE UPGRADE

- The line “Active Devices” indicates which of the selected remote amps are still to be updated. Amps for which the update process timed out are taken off the list. These devices are still capable of receiving upgrade commands. However, the software does not wait for acknowledgements of the concerned amps any longer.
- If the IRIS-Net-software recognizes an error or “Time Out” during upgrading, it automatically switches to “Single Step” mode, which offers the possibility to repeat the upgrade in single steps. If a “Time Out” message is displayed while upgrading is in progress, under no circumstance switch off any amps!
- As soon as “Single Step” is checked off, all buttons below the single step field become active. The upgrade can now be performed manually, step- by-step in the sequence as described below. If one of the commands does not finish “ok”, you have to restart the upgrade procedure from the beginning.

Step	**Description**
Start Update	Activates update mode for the selected devices. The messages window shows “Update started (addresses)” .. and after a short period of time “ok”.
Verify	Compares the firmware installed in the remote amps with the selected firmware file. The messages window shows “Verifying (addresses)”...a progression-bar indicates the approximate duration of the process. Detected differences are indicated at the end of the process, e.g. “done, Errors detected for...”. If no errors time-outs are detected, you can proceed with the update.
Erase Flashes	Deletes the actual firmware and clears the FLASH-memory of a remote amplifier. The messages window shows “Erasing (addresses)”..... and after a short period of time “ok”.
Program	Loads the new firmware into the FLASH-memory of a remote amplifier. The messages window shows “Programming (addresses)"........ A progression-bar indicates the approximate duration of the programming. “ok” appears in the message window after some time.

Checksum	Evaluates the checksum of the newly installed firmware. The messages window shows “Checksum (addresses)” .. and after a short period of time “ok”. This is a short form of the “Verify” process.
Stop Update	Finishes upgrading. The messages window shows “Finishing (addresses)" .. and after a short period of time “ok”. The remote amps quit the update mode and start in normal mode. Now, you can exit the upgrade dialog or proceed with upgrading other remote amps.

– If “Time Out” errors still occur during the programming, repeat the procedure in single step mode in the following sequence: Start Update - Pro- gram.
– If the checksum evaluation shows errors, repeat the entire upgrade procedure. Don’t forget to uncheck “Single Step” mode, for the upgrade to run automatically.

RCM-26

Using RCM-26 remote amplifiers

The IRIS-Net software (Intelligent Remote & Integrated Supervision) runs under Windows and allows configuring, controlling and monitoring a complete PA-system from a single or from several PCs. Any operational status, e.g. power-on, temperature, level, limiting, activation of protections, deviation of the output impedance, etc., are centrally recorded and displayed, which offers the opportunity to react and interfere even before the occurrence of critical operational states. Programming automated actions that are carried out when exceeding or falling short of certain threshold values is possible as well. All parameters, e.g. power-on/off, level, mute, filters, etc. are controlled in real-time and can be stored in any power amplifier.

Monitoring the connected loudspeaker systems is performed by continuously measuring output currents and voltages of individual power amplifier channels. Each exceeding or falling short of set thresholds is instantly signaled and logged. In this way, short-circuits or line interruptions, as they might occur during normal operation, are recognized and displayed immediately. The integrated impedance test function allows checking the connected loudspeaker systems more precisely. Together with the current/voltage testing function the integrated sweep-generator is employed to measure the connected loudspeakers' and cables' impedance over the entire frequency range. The resulting impedance graph is displayed on the PC-screen. Comparing the measured impedance progression with a reference value is possible at all times, which allows recognizing even the slightest defects or irregularities of the loudspeaker systems.

Besides controlling and monitoring amplifiers, the RCM-26 also offers all conventional signal processing functions, like parametric equalizers, frequency crossovers, delays and limiters. Beyond that, FIR-filters are available to optimize the amplifiers and loudspeaker system. All DSP-settings can be freely edited and stored in user presets directly on the module. Independent from network control all DSP settings (filter, delay, level) are maintained incase of failure.

Furthermore, the RCM-26 provides a control port with freely programmable control inputs and control outputs. Control inputs (GPI's) allow the connection of switches. IRIS-Net offers the possibility to program a variety of logic functions for the inputs (e.g. switching to an alarm-preset with maximum energy in the speech area). Control outputs (GPO's) allow the connection of external components, which, for example, are used to signal specific states to peripheral equipment. Consequently, an amplifier with a RCM-26 module installed corresponds to highest safety requirements.

The RCM-26 has been designed with uncompromising audio quality in mind. Analog audio inputs (internally) and an AES3 (AES/EBU) digital audio input with XLR-type connector are provided. The use of the digital audio input offers a dynamic range of 128 dB. Using the analog audio input offers a dynamic range of 120 dB, which, by the way, is an absolute peak value for digital audio devices.

Remote Amplifiers

The Remote Control Module RCM-26 can be used in following power amplifiers:

DYNACORD POWERH SERIE

- H 2500 2 x 1450 W / 4 Ohm or 2 x 2000 W / 2 Ohm
- H 5000 2 x 2500 W / 4 Ohm or 2 x 3500 W / 2 Ohm

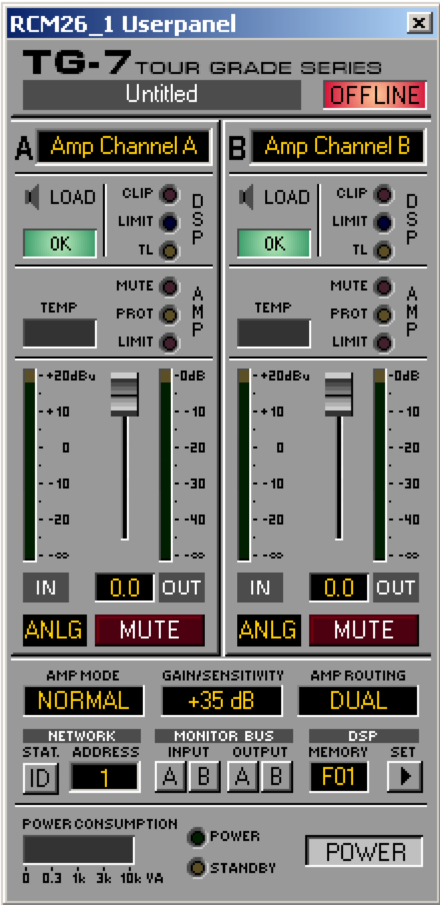
ELECTRO-VOICE TOUR GRADE SERIE

- TG-5 2 x 1450 W / 4 Ohm or 2 x 2000 W / 2 Ohm
- TG-7 2 x 2500 W / 4 Ohm or 2 x 3500 W / 2 Ohm

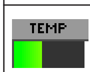
The power amplifiers marks a milestone in the design and the production of high-performance power amplifiers. The innovative 3-stage Grounded Bridge Class H Topology with ”floating” switching power supply unit offers very high and stable output with extreme high efficiency on an extremely high performance level at minimum weight.
PowerH / Tour Grade Series amplifiers are ideal for driving professional touring, high-end Concert-Sound and Pro-Sound applications. Next to classical protections, this new design employs the multi-stage ATP system (Advanced Thermal Protection) for the first time, which in most cases prevents the power amplifier from switching off when the temperature exceeds a critical level. The newly designed MCS system (Mains Current Supervision) prevents power amplifier breakdown caused by the activation of the automatic circuit breaker. For this, among other things, the MCS system uses the highly precise measurement of the RMS value of the actual mains current consumption. Information about the status of the power amplifier and its internal protections is provided on a LC-display. By utilizing the optionally available remote control module that is compatible with IRIS-Net, this power amplifier additionally offers comprehensive remote monitoring and remote control functions plus a universal 2-channel digital audio controller (DSP) including highly precise FIR-filtering.

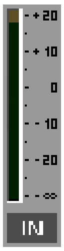
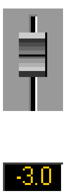
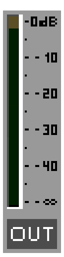
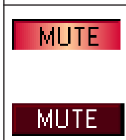
Amplifier Control Panel

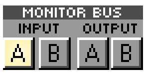
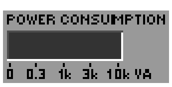
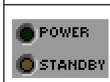
Double clicking with the left mouse button on an amplifier gets you to the Amplifier Control Panel, which provides access to the most important controls and indications of the selected amplifier. Simultaneously opening several Amplifier Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.



Element	Description
P 900 RL	Amplifier Type (generated during amplifier selection or read from the amp while being on-line)
☒	Using the left mouse button, click on the Close button to close the Amplifier Control Panel.
Stage Left	A name can be assigned to each amplifier to specify its use or position. Click on the gray-shaded entry field below the Amplifier Type field and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: Entering amplifier names is also possible within the Setup & Control Panel on the Config & Info page.** **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
ONLINE OFFLINE	The Online / Offline indicator signals whether the selected amplifier is included in the network or off-line. The red OFFLINE indicator signals that the corresponding amplifier is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding amplifier is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.

	The amplifier channels are named channel A and B. A name can be assigned to each channel to easily identify its allocation and use. Using the left mouse button, click in the entry field and enter the desired name for the channel. Press Return on the keyboard to acknowledge your entry. **HINT: Entering channel names is also possible within the Setup & Control Panel on the Config & Info page.**
	The CLIP indicator lights whenever the signal of the internal signal processor clips. The signal processor’s headroom is 2 dB, which is no problem when using normal filter settings. However, when drastically increasing the level of several adjacent or overlapping filters, distortion of high-level signals may occur, which the CLIP indicator indicates. In that case reducing the signal-level or trying a bit more moderate equalizer setting is recommended.
	The LIMIT indicator lights whenever the digital limiter of the corresponding channel is activated, e.g. when the signal level exceeds the specified threshold and the output level is being limited to this value.
	The TL indicator lights when the thermal limiter of the corresponding channel is activated.
	The LOAD indicator shows whether the load connected to the amplifier output is within the allowable range or if short-circuit or line interruption has occurred. The green OK-indication signals that the connected load is between the specified lower and upper limit values. These values are set in the Setup & Control Panel in the Load screen. The red OPEN indication signals line interruption. It lights whenever the connected load exceeds the upper limit value. The red SHORTED indication signals short-circuit at the amplifier output. It lights whenever the connected load falls below the lower limit value. **HINT: The connected load is monitored continually as soon as a signal with a voltage of > 150 mV is present at the output. Calculation of signal levels below that threshold is not possible and the indicator shows the last acquired state.**
	The TEMP display shows the amplifier’s internal temperature as a graph. The indicator lights green whenever the amplifier is operated in its normal operational temperature range. The indicator lights yellow whenever the amplifier builds up heat because of continuous high output. However, since the internal fans provide sufficient ventilation there is no risk of thermal overload in this state. As soon as temperature indication changes to red, reducing the output level is strongly recommended. Otherwise the amplifier might cease operation because of thermal overload.
	The MUTE indicator lights when the amplifier is muted. This occurs e.g. during speaker switch-on delay or switching the amplifiers input sensitivity.
	When the red PROT indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated However, a lit PROT LED does not necessarily mean that the signal path gets switched off. The differentiated protections concept of the power amp results in several protection circuits being activated one after another, which ensures that under normal circumstances the power amplifier will stay in the safe and stabile operating range. In case the amplifier needs to be switched off to prevent power amplifier and connected speaker systems from being damaged, this is indicated by the PROT and MUTE LEDs being lit simultaneously.

	The LIMIT indicator lights as soon as the internal dynamic limiter is activated, which is the case when the amplifier is operated at maximum out- put. Short-term blinking is not a problem, since the internal limiter controls input levels of up to +20dBu down to a distortion rate of approximately 1%. However, if this indicator lights permanently, reducing the output level is strongly recommended to protect the connected loudspeaker systems from being damaged by capacity overload.
	The Input Level Meters provide indication of the corresponding audio levels at the amplifier inputs in dBu. The amplifier’s nominal input level is +6dBu, the maximum level can be as high as +21dBu. In general, it is recommended that the amplifier be operated in a range between 0 and +10dBu. Only signal peaks should be at higher levels.
	The level controls are for adjusting the overall amplification of the corresponding amplifier channel. Setting the level controls to a value between 0dB and -6dB provides full output capacity. The numerical field below the level controls indicates the set level, by which the output amplification is attenuated, in dB.
	The Output Level Meters provide indication of the corresponding audio levels at the amplifier outputs. Indication in dB is relative to amplifier full-modulation. A 0dB output level (full-modulation) is indicated in yellow.
	The currently used audio input (ANLG or AES3) is indicated.
	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	AMP MODE indicates the operation mode of the power amplifier blocks. Possible settings are NORMAL and BRIDGED. Switching the amp mode is only possible locally at the power amplifier, details can be found in the amplifiers owner's manual.
	GAIN/SENSITIVITY displays the amplifiers constant gain of +35 dB.
	AMP ROUTING shows how the audio inputs handle the input signals. Possible settings are DUAL and PARALLEL. Switching the amp routing is only possible locally at the power amplifiers, details can be found in the amplifiers owner's manual.
	Clicking this switch activates the STATUS indicator on the amplifier’s rear panel as well as in the amplifier’s front panel window in the IRIS-Net software. Normally, the STATUS indicator blinks only during serial communication. Once the STATUS switch is engaged, the STATUS indicator blinks in a steady but fast sequence. This function is meant for checking communication and for identifying or searching an amplifier in a large system setup.

	The address field indicates the set amplifier address. Assigning a new address is also possible by clicking into the field with the left mouse but- ton and entering the desired amplifier address. Available values are 1 to 250. Press Return on the computer keyboard to acknowledge your entry. The assigned address and the address specified by the setting of the selection switch on the amplifier’s rear panel have to be identical. Each address can exist only once within a system.
	These buttons allow assigning amplifier channels to the monitor bus. The monitor bus allows monitoring any amplifier input or output signals within an installation. INPUT A / B selects the corresponding input signal while OUTPUT A / B allows switching between the output signals of channels A and B. Simply click on an amp channel’s icon to select it for monitoring. The corresponding channel is assigned to the monitor bus. Any previous selection is simultaneously canceled, so that only the actually selected amp channel can be monitored. Clicking the button of an active amp channel separates the channel from the monitor bus.
	This field indicates the active factory, user or owner preset. Each remote amp has two factory setting F01 (48 kHz) and F02 (96 kHz) offering linear settings and six user-programmable presets U01...U06 for storing random user data. There are also two password-protected owner presets. Loading and saving presets is done in the Setup & Control window.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all amplifier- and DSP-parameters, control and monitoring functions plus additional function groups.
	POWER CONSUMPTION indicates the current power consumption of the power amplifier in VA.
	This soft-key allows switching an amplifier on or off. The STANDBY and POWER indicators signal the actual operational status. The Config & Info window allows programming individual power-on delays for each amplifier. **HINT: The power-on delay defaults to “Address*150ms”. For address 8 the power-on delay default would be for example: 8*150ms=1200ms.**
	These indicators show the amp’s actual operational status. STANDBY lights whenever the amplifier is in stand-by mode. POWER lights whenever the amplifier is powered-on and ready for operation. If neither one of the indicators lights, the amplifier is either off-line or powered-off.

Setup & Control

The Setup & Control window allows configuring all amplifier parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

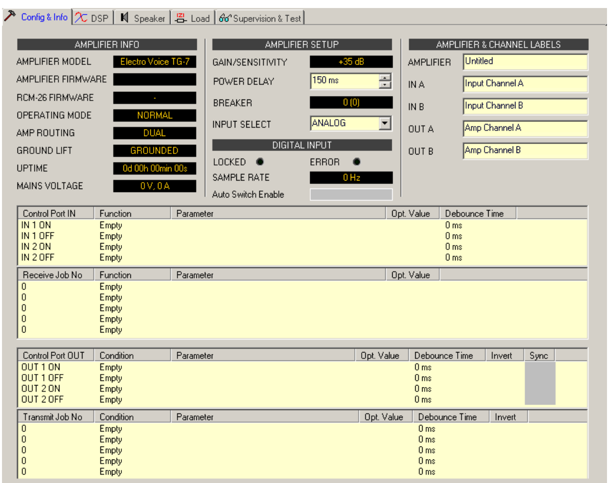
Window	**Description**
Config. & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Filter, Delay, X-Over, Limiters) of the amplifier.
Speaker	This page allows loading and displaying speaker data.

Load	This page provides access to several settings for impedance/load monitoring and impedance testing.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.

Clicking on the „SET“ soft key in the Amplifier Control Panel opens the Setup & Control window.

Config & Info

The Config & Info window provides information and basic settings for the selected amplifier. Additionally, editing labels is possible in this window.
To select the page click on the Configuration & Information tab in the Setup & Control Window.



Amplifier Info

Element	Description
AMPLIFIER MODEL	Shows the amplifier type
AMPLIFIER FIRMWARE	Shows the amplifier‘s software version number (operating system, firmware)
RCM-26 FIRMWARE	Shows the remote control module‘s software version number (operating system, firmware)
OPERATING MODE	Shows the amplifier’s operating mode. The power amplifier can be operated in NORMAL or BRIDGED mode.
AMP ROUTING	Shows how the amplifier's audio inputs handle the input signals. Possible amp routings are DUAL and PARALLEL.
GROUND LIFT	Shows the setting of the amplifier's GROUND LIFT switch. Possible settings are GROUNDED and UNGROUNDED.
UPTIME	Shows the uptime of the amplifier (standby not included) since the last reset of the Event Log. For further details about the amplifier’s Event Log please refer to the owner's manual.
MAINS VOLTAGE	Shows the mains voltage and the mains current consumption.

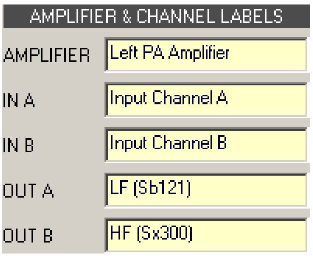
Amplifier Setup

Element	Default	Range	Description
GAIN/ SENSITIVITY			GAIN/SENSITIVITY displays the amplifiers constant gain of +35 dB.
POWER DELAY	Address * 150 ms	50...4000 ms 50 ms Steps	Allows programming an amplifier’s power-on delay. Setting different delay times is recommended to prevent the mains fuse from blowing when powering on several amps at the same time.
BREAKER			Show the current setting of the amplifier's Mains Circuit Breaker Protection. For further details about this protection please refer to the owner's manual.
INPUT SELECT			Allow selecting the audio input. The analog audio input (ANALOG) or the digital audio input (AES/ EBU) are available. **HINT: The value of the property “Audio Input” corresponds to the currently used audio input. You can write the value “ANLG” to this property to select the analog audio input. You can write the value “AES” to this property to select the digital audio input.**

Digital Input

Element	Default	Range	Description
			Indicates if the digital audio input is synchronized to the input signal (LOCKED LED on) or if synchronization was not successful (ERROR-LED off). **HINT: The current state of the digital audio input is available via the property “StateFlags. AES Locked” also.**
SAMPLE RATE	96 kHz	32...129 kHz	Shows the sample rate of the digital input signal.
Auto Switch Enable			Activate this control if the audio input should switch from digital input to analog input automatically if the digital input signal is not OK. **HINT: The value of the property “Audio Input Switching. AES Fail. Enable” corresponds to this control.**

Amplifier & Channel Labels

Element	Description
	The labels of an amplifier and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately reflected in the different panels and windows (amplifier control panel, flow diagram, and overview). **CAUTION: Using*(asterisk) and/or = (equal) signs in a name is not permissible.**

SWITCHING AUDIO INPUTS AUTOMATICALLY

RCM-26 remote amplifiers allow automatic switching from the AES feed to the analog input in the case of failure of the AES signal or input. This feature allows a redundant analog signal to be fed to the amplifier and automatically switch over to the “backup” signal without user intervention.

Open the Modify Properties dialog from the context menu of the RCM-26 remote amplifier in the IRIS-Net worksheet. Following table lists the relevant properties for automatic audio input switching.

Property	Default	Range	Description
Audio Input Switching. AES Fail. Enable	0	0, 1	Set the value to 1 to enable the auto fall back function which switches from AES to analog input in case of an AES error. This property corresponds to the „Auto Switch Enable“ control found in the Config & Info window.
Audio Input Switching. AES Fail. Time	1 s	0 to 120 s	Configures the duration the AES error had to be present on the input to trigger the AES / analog switching.
Audio Input Switching. AES Good. Enable	0	0, 1	Setting this property to 1 enables the automatic switching from analog input to AES input in case the AES signal is OK. **HINT: This works only if the values of both properties – Audio Input Switching. AES Fail. Enable and Audio Input Switching. AES Good. Enable-are set to 1.**
Audio Input Switching. AES Good. Time	5 s	0 to 120 s	Configures the duration the AES signal had to be in „Locked“ state, to switch back from analog to the AES input.
Audio Input Switching. AES Ok Flag	0	0, 1	Shows that the AES signal is locked and without errors for longer than the configured Audio Input Switching. AES Good Time.
Audio Input Switching. AES Selected Flag	0	0, 1	Shows that the AES input is used as amplifier audio input.
Audio Input Switching. Counter	0	-	Counts how often the Auto Switch ganged the input.

Control Port

A control port offering two control inputs and two control outputs is located on the amplifier's rear panel. The functions of these inputs and outputs can be programmed in a variety of ways. For example, the control inputs (GPI) can be used for power-on / stand-by or preset switching as well as for changing parameters. The control outputs (GPO) are for signaling internal statuses. They can directly trigger LEDs, control lamps or relays. In the Supervision & Test window the states of the control inputs are displayed and you have the possibility to switch the control outputs manually. For more information and electrical specifications of the control port, please refer to the amplifier's manual.

Control Inputs: Each status change of a control input can trigger a function. Different functions can be assigned for the opening (OFF) or closing (ON) of a contact.

Example:

Control Port IN	Function	Parameter	Opt. Value	Debounce Time
IN 1 ON	Power	on		0 ms
IN 1 OFF	Power	off		0 ms
IN 2 ON	Preset	U03		0 ms
IN 2 OFF	Preset	U02		0 ms

This example shows the programming of two control inputs where IN1 switches the amplifier on or off and IN2 selects presets U02 or U03.

- IN1 ON: Power on (closing the contact of control input 1 switches the amplifier on)
- IN1 OFF: Power off (opening the contact of control input 1 switches the amplifier to stand-by)
- IN2 ON: Preset U03 (closing the contact of control input 2 selects preset U03)
- IN2 OFF: Preset U02 (opening the contact of control input 2 selects preset U02)

Element	Default	Range	Description
Control Port IN		IN 1 ON IN 1 OFF IN 2 ON IN 2 OFF	This provides a listing of the two control inputs and their statuses ON and OFF. The entries in the corresponding lines specify the action when closing (ON) or opening (OFF) a contact.
Function	(empty)		This column allows assigning functions to a control input's statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. Following a status change the assigned function is initiated after the set time interval has past.

Control Outputs: Internal status changes inside of the amplifier, such as operational faults, alerts when exceeding parameter limits, and internal operational statuses can be signaled to external systems or central control units.

Example:

Control Port OUT	Condition	Parameter	Opt. Value	Debounce Time	Invert	Sync
OUT 1 ON	Power			0 ms		
OUT 1 OFF	Power			0 ms	X	
OUT 2 ON	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms		
OUT 2 OFF	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms	X	

This example shows the programming of two control outputs where OUT1 signals whether the amplifier's power is switched ON or OFF while OUT2 signals faulty operation.

– OUT1 ON: Power (control output 1 is closed when the amplifier's power is switched on)
– OUT1 OFF: Invert Power (control output 1 is open when the amplifier's power is switched off / stand-by mode)
– OUT2 ON: Errorflag (control output 2 is closed when operational faults according to the parameter list have occurred)
– OUT2 OFF: Invert Errorflag (control output 2 is open when no faults have occurred)

Element	**Default**	**Range**	**Description**
Control Port OUT	0	OUT 1 ON OUT 1 OFF OUT 2 ON OUT 2 OFF	This provides a listing of the two control outputs and their statuses ON and OFF. The entries in the corresponding lines specify which status results in the closing (ON) or opening (OFF) of a contact.
Condition	(empty)		This column allows assigning internal events (conditions) to a control output's statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual set- tings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. An event is signaled following an internal status change and after the specified time interval has past.
Invert	(empty)	(empty) / X	This column allows entering whether a status is signaled when the specified Condition is „true“ (no entry) or „false“ (click „X“ to signal an inverted state).
Sync	(empty)		This column displays the SYNC flag. „X“ specifies that the output is synchronized with a sync- signal. This flag is erased when entering a new Function.

Jobs

For amplifiers to be able to communicate with each other, it is possible to send and receive Job Codes. In principle, a job code is a function number that an amplifier transmits via CAN-bus and that is received and interpreted by another or several other amplifiers. Each amplifier is capable of transmitting and receiving up to 5 different job codes. Programming job codes is nearly identical to the programming of control inputs and outputs.

Receive Jobs: A receive job is a function that is carried out as soon as the corresponding function number (the Receive Job Code) is received.

Example:

Receive Job No	Function	Parameter	Opt. Value
1	Power	on	
2	Power	off	
3	Preset	U03	
4	Preset	U02	
0	Empty		

This example shows the programming of four Receive Jobs. Jobs No. 1 and 2 switch the amplifier's power on or off while jobs No. 3 and 4 select presets U03 or U02. The fifth Receive Job has not been configured.

– Receive Job Nr. 1: Power on (receiving Job Code 1 switches the amplifier's power on)
– Receive Job Nr. 2: Power off (receiving Job Code 2 switches the amplifier into stand-by mode)
– Receive Job Nr. 3: Preset U03 (receiving Job Code 3 selects preset U03)
– Receive Job Nr. 4: Preset U02 (receiving Job Code 4 selects preset U02)

Element	Default	Range	Description
Receive Job No	0	1...1023	Here, you can specify which incoming job code numbers a specific amplifier recognizes. Entering random numbers between 0 and 1023 is possible.
Function	(empty)		This column allows assigning an individual function to each job code received. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.

HINT: Programming identical control functions or receive jobs for several amps is easily accomplished by creating Programming identical control functions or receive jobs for several amps is easily accomplished by creating a group that includes all the desired amps and afterwards perform the programming in the group's Configuration& Information dialog. All settings are automatically applied to all amplifiers of that group, which saves time and effort and additionally reduces the risk of programming errors.

Transmit Jobs: Transmit Job defines a function number that is sent as soon as a specific internal event (condition) occurs in the amplifier.

Example:

Transmit Job No	Condition	Parameter	Opt. Value	Debounce Time	Invert
1	GPI	IN1		0 ms	
2	GPI	IN1		0 ms	X
3	GPI	IN2		0 ms	
4	GPI	IN2		0 ms	X
0	Empty			0 ms	

This example shows the programming of four Transmit Jobs. Jobs No. 1 and 2 are triggered by control input 1. Jobs No. 3 and 4 are triggered by the status signaled from control input 2. The fifth transmit job has not been configured.

– Transmit Job Nr. 1: GPI IN1 (Job Code 1 is transmitted when control input 1 is closing)
– Transmit Job Nr. 2: Invert GPI IN1 (Job Code 2 is transmitted when control input 1 opens)
– Transmit Job Nr. 3: GPI IN2 (Job Code 3 is transmitted when control input 2 is closing)
– Transmit Job Nr. 4: Invert GPI IN2 (Job Code 4 is transmitted when control input 2 opens)

Element	Default	Range	Description
Transmit Job No	0	1...65536	Here, you can specify which job code numbers an amplifier transmits on the occurrence of specific events. Entering random numbers between 0 and 65536 is possible.

Condition	(empty)		This column allows specifying an event (condition) that triggers the corresponding transmit job code. Clicking the desired line in the Condition menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here, you can program delay or debouncing times. A transmit job code is sent following a specific event and after the specified time interval has past.
Invert			This column allows entering whether a job code is transmitted when the specified Condition is "true" (no entry) or "false" (click "X" to signal an inverted state).

Input and Receive Job Function: The following table lists all functions together with their individual settings, which can be triggered via control input or Receive Job.

Function	**Parameter**	**Opt. Value**	**Function executed**
Empty	-	-	None
Power	off on flip		Power Off (Standby) Power On Power-status change (ON to Stand-by and reverse)
Absolute	All DSP parameters	Corresponding Parameter Value (parameter-dependent)	Set the specified absolute parameter value for the selected parameter
Relative	All DSP parameters	Parameter Value Off- set (parameter-dependent)	Changes the actual value of the selected parameter by the specified offset value
Flip	Parameters with two statuses		Changes the status of the selected parameter (e.g. bypass On / Off)
Preset	U01 - U08, F01		Changes a preset to the specified preset number
Monitor	Relay, IN A, IN B, OUT A, OUT B	on, off	Activates respectively deactivates the selected monitor bus signal
Memo flag	Set, Clear, Toggle Memo flags 1 - 16		Sets, erases or changes selected memory flags. Up to 16 memory flags are available and simultaneously accessible.

Measurement	Generator frequency, Time, Level A / B		Starts measurement with a tone signal of the specified frequency at the levels specified for channels A / B for the selected duration (0 ms = infinite)
Test generator	Channel, Signal type, Frequency, Solo/Pre, Level		Starts the test generator with selected signal type or of the specified frequency at the levels specified for channels A / B for the selected duration (0 ms = infinite)

Output and Transmit Job Conditions: The following table lists all amplifier statuses that can be used for triggering control outputs or for sending Transmit Job Codes.

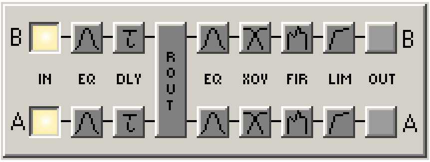
Function	Parameter	Opt.Value	Invert	Triggering Event/Status Change
Empty	-	-		Not configured
Power			X	Power On Power Off (Standby)
Absolute	all DSP parameters	Corresponding Parameter Value (parameter-dependent)	X	Set parameter value reached or exceeded Set parameter value declined
Temp	Temperature in °C		X	Set temperature reached or exceeded Set temperature declined
VU	IN A, IN B, OUT A, OUT B, Limiter A/B, Compressor A/B	Level in dB	X	Set level reached or exceeded Set level declined
GPI	IN 1, IN 2		X	Control input 1 / 2 closed (ON) Control input 1 / 2 open (OFF)
Errorflag	All internal fault conditions		X	Single or several error flags set None of the selected error flags set
Memoflag	Enable for selected flags as well as bit- pattern of flags 1 - 16		X	Memory flags match the selected bit-pattern Memory flags do not match the selected bit-pattern
Preset	U01 - U08, F01-F02, O01-O02		X	Specified preset selected Other than the specified preset selected

DSP

The DSP pages provide overview and access to all DSP parameters of an amplifier. Within this window you can use the Flow Diagram Selector to link to different function groups.

FLOW DIAGRAM SELECTOR

The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.



A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

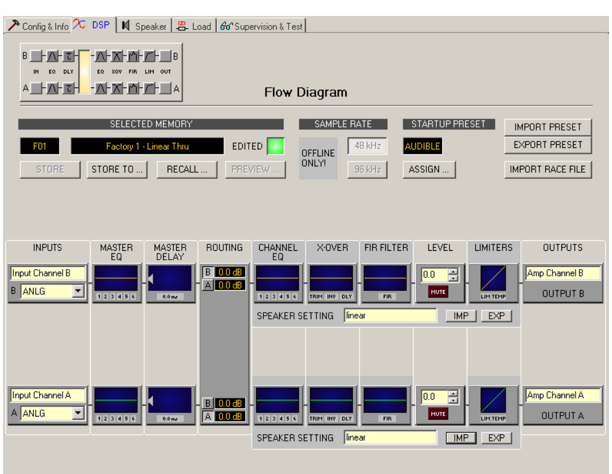
Selte	**Description**
Flow Diagram	The signal flow display provides an overview of an amplifier's DSP settings. This area also includes all controls for the preset location and preset file management.
Master EQ	The MASTER EQ page provides access to the two 6-band parametric equalizers of the amplifier inputs.
Master Delay	This page allows the programming of delay lines for the amplifier channels A and B.
Channel EQ	The CHANNEL EQ page offers access to the two 6-band parametric equalizers of the amplifier outputs for speaker equalization.
X-Over	Frequency crossover-filters as well as the parameters gain, polarity and alignment-delay for both channels are located in the X-OVER area.
FIR-Filter	This page provides a FIR-Filter for each amplifier channel.
Limiters	This page provides access to Peak Anticipation limiter and Thermal limiter of each amplifier channel.

The DSP functions of a remote amplifier can be accessed by clicking onto the SET key in the Amplifier Control Panel followed by a click on the DSP register in the Setup & Control Window.

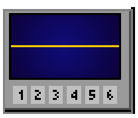
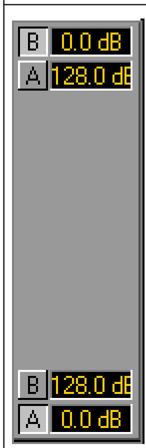
FLOW DIAGRAM

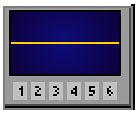
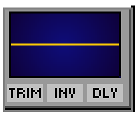
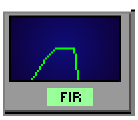
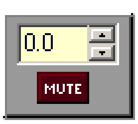
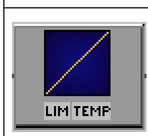
The FLOW DIAGRAM window shows a signal flow diagram, which offers a quick overview of all DSP setting of an amplifier. Labeling and routing channels can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for the saving, loading and previewing of loudspeaker presets are also accessible from this window.

The FLOW DIAGRAM window opens when clicking on the first, fourth or ninth block in the Flow Diagram Selector.



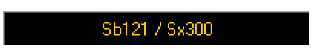
Function blocks

Element	Description
	Input Block: The text field allows specifying a name for the corresponding input channel. The drop down allows selecting the audio input type of the corresponding input channel. **HINT: Selecting different audio input types for the amplifier's input channels is not possible.** A click with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding input channel (Master EQ, Master Delay) to any other input channel within the same project.
	Master EQ Block: The Master EQ block displays the 6 Master EQs of the corresponding input channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Master EQ block. A single click with the left mouse button onto this block opens the MASTER EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Master Delay Block: This displays the Master Delay of the input channels. The corresponding LED signals whether a delay has been programmed or not. The delay-value is displayed together with the measurement unit next to the LED. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the MASTER DELAY page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Master Delay block to any other Master Delay block within the same project.
	Routing Block: Here you can assign the output channel routing. The A and B buttons allow selecting the input signal for the corresponding output channel. Clicking with the right mouse button onto the dB display opens a fader. A click with the right mouse button onto routing block opens the Copy & Paste menu of all DSP set- tings, which allows copying all DSP parameters of an amplifier to any other amplifier within the same project.

	Channel EQ Block: The Channel EQ block displays the 6 Channel EQs of the corresponding output channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Crossover Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse button onto this block opens the X- OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X-Over block within the same project.
	FIR Filter Block: This block represents the FIR Filter within the corresponding output channel. The graph shows the frequency response that results from the set FIR parameters. The LED indicate if the FIR Filter is being used. A single click with the left mouse button onto this block opens the FIR page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding FIR Filter block to any other FIR Filter block within the same project.
	Level Block: The numerical field is identical to the numerical field below the level controls in the Amplifier Control Panel. So the field indicates the actually set attenuation, by which the internally specified amplification is attenuated, in dB. The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Limiters Block: This block provides graphical display of the limiter functions of the corresponding output. The two LEDs indicate whether peak limiter or thermal limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Limiters block to any other Limiters block within the same project.

	Allows import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for DSP blocks Channel EQ, Crossover, FIR Filter and Limiters. The text field allows editing the description of the Speaker Setting file.
	Output Block: The text field allows specifying a name for the corresponding output channel. A click with the right mouse button onto OUT A or OUT B opens the Copy & Paste menu, which allows copying all parameters of the corresponding output channel (Routing, Channel EQ, X-Over, FIR Filter, Dynamics) to any other output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied.

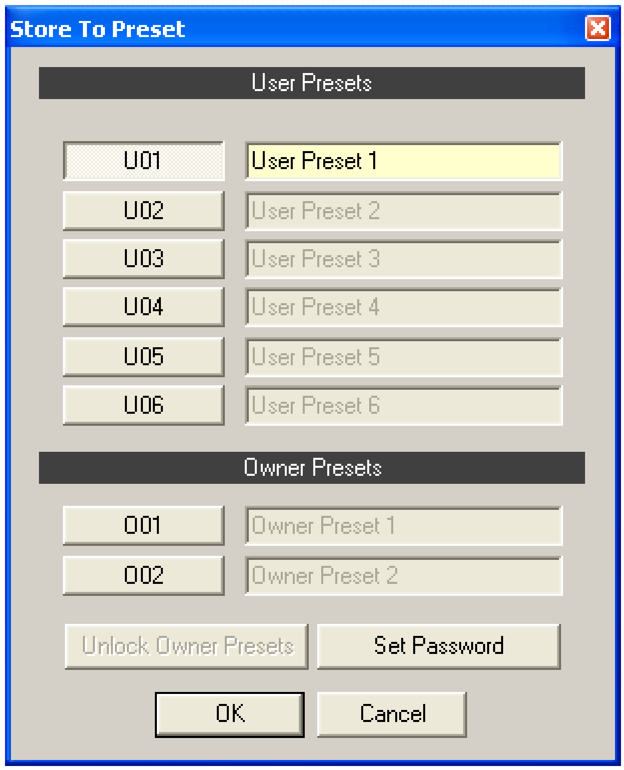
Status Indication

Element	Description
	The MEMORY display shows the number according to the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	NAME indicates the name of the actually audible preset.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

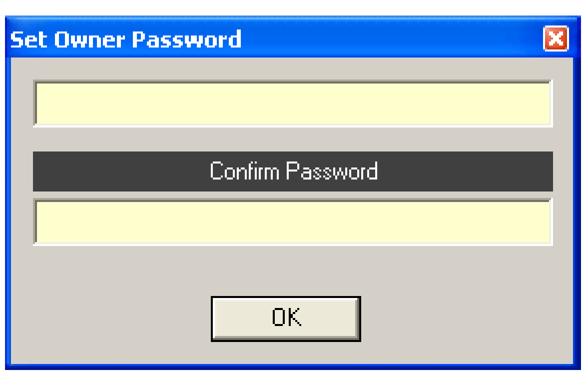
Store / Recall / Preview

Element	Description
	STORE saves all momentary set DSP parameters into the actually loaded preset.

<table>
<tr>
<td>
</td>
<td>STORE TO...saves all momentary set DSP parameters into a selectable User Preset. In online mode the parameters can be saved into a selectable Owner Preset. A click with the left mouse button opens the "Store To Preset" dialog, where the preset can be selected and a name can be assigned to the preset.



In online mode the password protection of the owner presets can be activated. Press the Set Password button to open the „Set Owner Password“ dialog.



Enter the password into the text boxes and confirm the password by pressing the OK button. For editing protected Owner presets press the Unlock Owner Presets button and enter the password.</td>
</tr>
</table>

RECALL	RECALL loads and displays all DSP parameters that are stored in the selected preset. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
PREVIEW	PREVIEW reads and displays all DSP parameters that are stored in the selected preset. This function is used to display and check a preset's contents, without actually loading the preset. You can neither listen to the preset nor edit its contents, as long as you do not explicitly load it using the RECALL function.

Preset after Startup

Element	Description
STARTUP PRESET AUDIBLE ASSIGN ...	After switching on the power amplifier a specified Preset can be loaded or the last DSP settings can be restored. Click the ASSING... button with the left mouse button opens the Assign Startup Preset dialog, where the Preset to be recalled or Audible for restoring last DSP settings can be selected.

Switching sample rate

Element	Description
SAMPLE RATE OFFLINE ONLY! 48 kHz 96 kHz	The RCM-26 Remote Control Module allows signal processing with a sample rate of 48 kHz or 96 kHz. **HINT: Switching sample rate is only possible in off-line mode.**

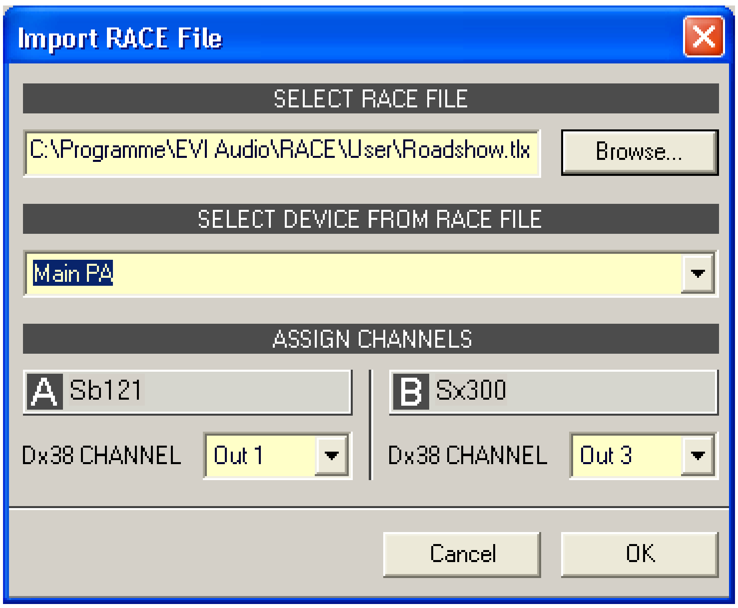
Import / Export of Preset Files

IRIS-Net allows the storing of all DSP parameters of an amplifier together with the according preset name in a file, and to load amplifier parameters from these files. Therefore, IRIS-Net creates a sub-directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory /Presets is permissible.

Element	Description
IMPORT PRESET	After clicking onto IMPORT PRESET appears an „Open File“ dialog box. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loud speaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a „Save File“ dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store all DSP parameters together with the corresponding file name. „.ds“ is automatically added as file extension.

Import of EV RACE Files

Element	Description
IMPORT RACE FILE	IRIS-Net allows importing loudspeaker presets that have been created in Electro-Voice RACE by clicking onto the button IMPORT RACE FILE to open the following dialog box.

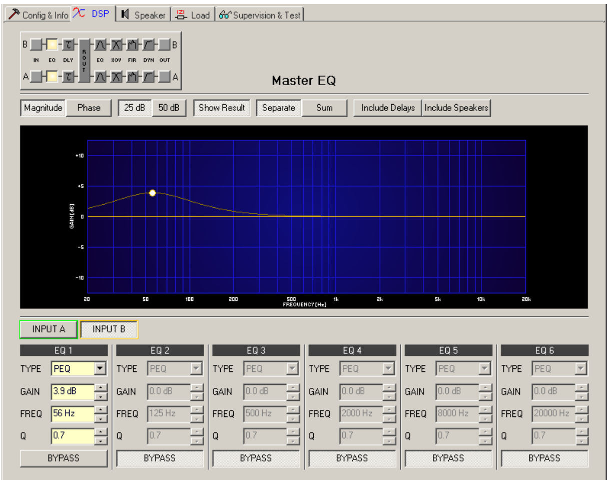


First, you have to select the desired RACE file by use of the Browse... button. Because a RACE file can hold the data of up to 31 EV Dx38, you need to continue by selecting the desired device from the RACE file within the dialog SELECT DEVICE FROM RACE FILE. At the end you have to specify which of the four Dx38 output channels should be assigned to the corresponding amplifier channels. Clicking onto OK button completes the process.

Caution!
The loaded RACE file becomes instantly audible when in on-line mode. Be sure to select the desired file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

MASTER EQ

Both input channels of a remote amplifier employ 6-band parametric equalizers each, which allow programming highly variable full-range speaker equalization to match a PA-system to different environmental and acoustical requirements. In many cases post-mixing console parametric equalization becomes redundant.
The Master-EQ is selected by clicking on the second block of the flow diagram selector or by double clicking on the MASTER EQ block in the full-scale flow diagram.



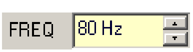
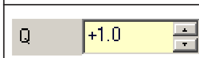
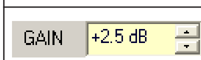
Graphics Display Indication

Element	Description
Magnitude / Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB / 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings – the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while “electrical” graphs are indicated in dark colors.
Separate / Sum	Selecting “separate” results in a separated display of the two amplifier channels’ transfer functions while “sum” shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the “Speaker” register sheet.

Channel Selection

Element	Description
INPUT A / INPUT B	Switch for selecting input A or input B for filter editing. A click with the right mouse button opens the “Copy & Paste” menu, which allows convenient copying all EQs of the corresponding input to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the “Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following edit- able parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
	31/ 125 / 500 /	20 Hz...20	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving
	2000 / 8000 /	kHz	and Hi / Lo pass filters.
	20000 Hz		
	0.7	0.4...40.0	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		(PEQ),	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the res-
		0.4...2.0 (Hi-/	ponse of Hi, Lo and All pass filters with slopes of 12dB/oct.
		Lopass)	
	0 dB	-18...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via “Mouse Movement” in the Graphics Display

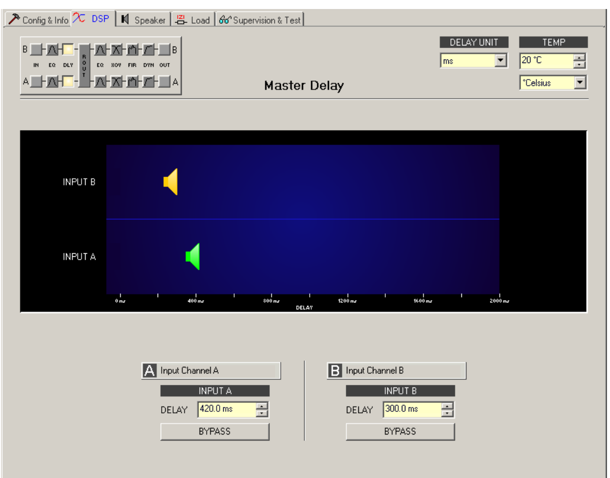
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its amplification (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

MASTER DELAY

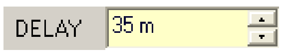
Individual master delays can be set for each input channel of a remote amplifier. Setting a different delay for the summed signal of the two input channels is also possible. Master Delays are mainly used to compensate for different natural delay times in the audio signal, as they are common when two sound sources reproducing identical audio information are located further apart.

You can select the master delay window by clicking onto the third block in the Flow Diagram Selector or by double clicking onto the MASTER DELAY block in the flow diagram.

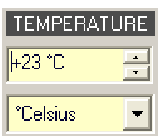


Channel Parameters

Element	Default	Range	Description
A Mix In			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all master delay parameters of the selected input to any other master delay within the same project.
INPUT A			Channel identification A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all master delay parameters of the selected input to any other master delay within the same project.

	0.0 ms	0.0...2000 ms	DELAY allows delaying the corresponding input channel’s audio signal by an adjustable period of time.
			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

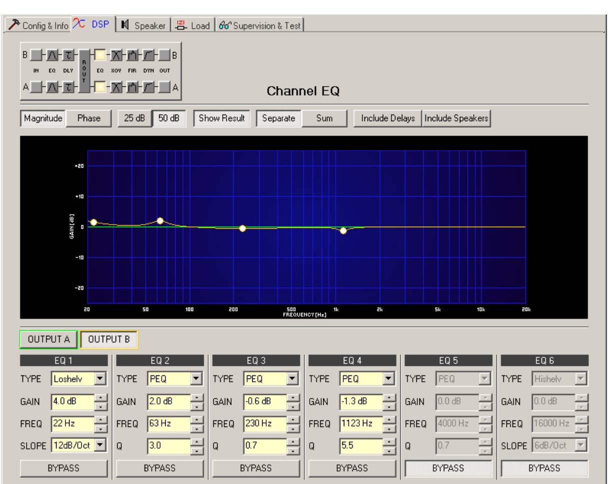
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 °C	-20...60 °C or -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by Dragging the Mouse in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title lights in color as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

CHANNEL EQ

Both output channels of a remote amplifier employ 6-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select All pass as filter type, these filters are identical to the ones of the master-EQ's. The Channel-EQ is selected by clicking on the fifth block of the flow diagram selector or by double clicking on the CHANNEL EQ block in the full-scale flow diagram.



Graphics display Indication

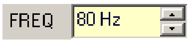
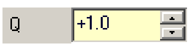
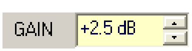
Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
Separate Sum	Selecting separate results in a separated display of the two amplifier channels' transfer functions while sum shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

Channel Selection

Element	Description
OUTPUT A OUTPUT B	Switch for selecting output A or output B for filter editing. A click with the right mouse button opens the Copy & Paste menu, which allows convenient copying all EQs of the corresponding output to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the „Copy & Paste“ menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE Hipass	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following edit- able parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.

	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
	20 / 63 / 250 /	20 Hz...20	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving
	1000 / 4000 /	kHz	and Hi / Lo pass filters.
	16000 Hz		
	0.7	0.4...40.0	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		(PEQ),	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the res-
		0.4...2.0 (Hi-/	ponse of Hi, Lo and All pass filters with slopes of 12dB/oct.
		Lopass),	
		0.4...2.0 (All-	
		pass)	
	0 dB	-18...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with All pass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

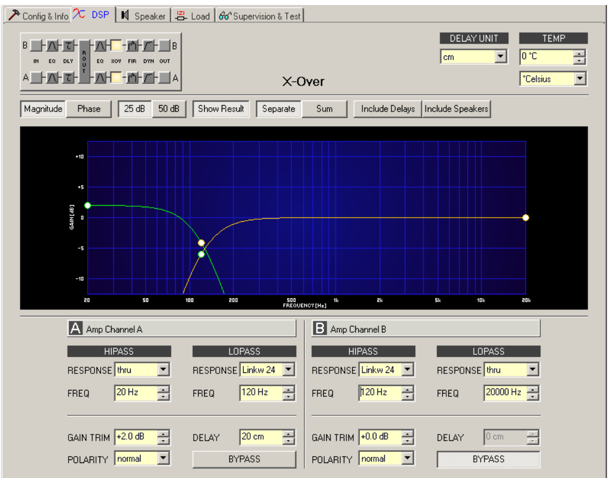
Filter Editing via Mouse Movement in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter's frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

X-OVER

The X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch, which are provided for each output channel of a remote amplifier. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.
Clicking on the sixth block in the Flow Diagram Selector or double clicking on the X-OVER block in the large signal flow diagram opens the X-Over window.

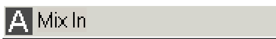
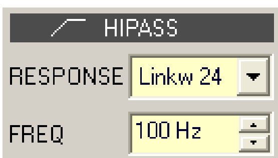
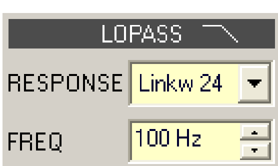


Graphics Display Indication

The graphics display offers several different indication modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (Master EQ, Channel EQ), which always provides precise over-view and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
Separate Sum	Selecting separate results in a separated display of the two amplifier channels’ transfer functions while sum shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels’ summed signal reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

Channel Parameters

Element	Default	Range	Description
			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all X-Over parameters of the corresponding output to any other X-Over within the same project.
	thru, 20 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/ Q=0.6, 12dB/Q=0.7, 12dB/ Q=0.8, 12dB/Q=1.0, 12dB/ Q=1.2, 12dB/Q=1.5, 12dB/ Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bes- sel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI-PASS filters within the same project.
	thru, 20000 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/ Q=0.6, 12dB/Q=0.7, 12dB/ Q=0.8, 12dB/Q=1.0, 12dB/ Q=1.2, 12dB/Q=1.5, 12dB/ Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bes- sel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.

GAIN TRIM +6.0 dB	0 dB	-30 dB...6 dB	GAIN TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
POLARITY normal	normal	normal, inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to Sum).
DELAY 15 cm	0.0 ms	0.0...350.0 ms	DELAY allows delaying the audio signal of the corresponding output by an adjustable period of time. This delay method is typically used as time-alignment-delay to overcome negative sound effects like they result from different distances between loudspeaker systems within one cabinet or the positioning of speakers in a PA-installation that otherwise would cause a high amount of natural delay.
GAIN +2.5 dB			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

Element	Default	Range	Description
DELAY UNIT m	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
TEMPERATURE +23 °C °Celsius	20 °C	-20...60 °C or -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

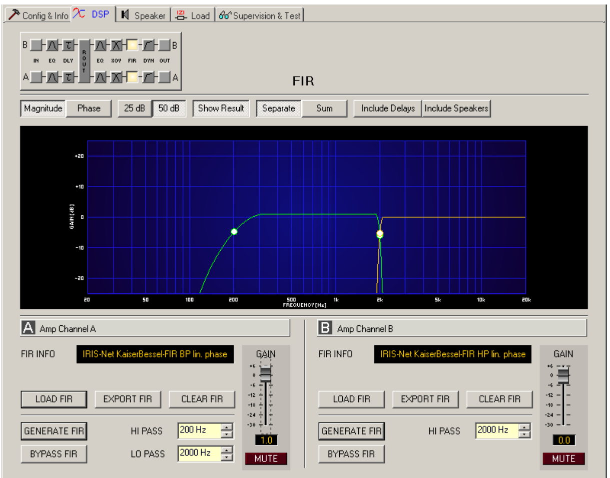
Editing X-Over Filters by Dragging the Mouse in the Graphics Display

Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right.
A filter’s title lights in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling. An extra white graph is displayed in addition, representing the frequency response of the corresponding selected filter.

FIR-FILTER

This page provides access to the two FIR filters in the amplifier's output channels. FIR filters can be generated from scratch, imported from files or exported to files for later use.

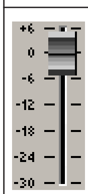
Clicking on the seventh block in the Flow Diagram Selector or double clicking on the FIR block in the large signal flow diagram opens the FIR window.



Graphics display Indication

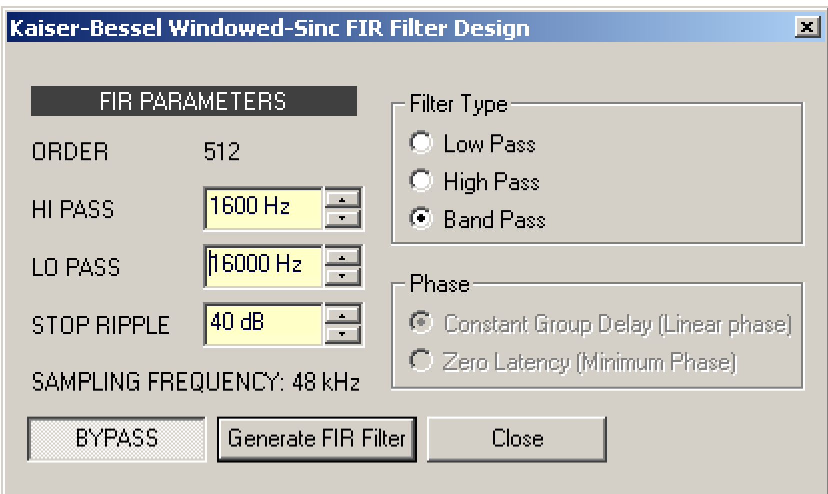
Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the amplifier outputs. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
Separate Sum	Selecting separate results in a separated display of the two amplifier channels’ transfer functions while sum shows the summed signal of the amplifier channels.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the amplifier channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

Channel Parameters

Element	Description
	Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all FIR filter parameters of the corresponding output to any other FIR filter within the same project.
	Description of the actually used FIR filters.
	After clicking onto LOAD FIR appears an "Open File..." dialog box. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
	After clicking onto EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
	Clears the actually used FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
	Clicking onto the GENERATE FIR buttons opens the Filter Design dialog.
	BYPASS switches the corresponding FIR filter ON (not engaged) or OFF (engaged), which allows for quick A / B- evaluation of the actual effect that a filter has on the sound.
	HI PASS sets the cut-off frequency of the Hi pass filter.
	LO PASS sets the cut-off frequency of the Lo pass filter.
	Allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB.

0.0	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
MUTE	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.

FIR Filter Design



Element	Default	Range	Description
ORDER 512			The ORDER is set to 512.
HI PASS 200 Hz	200 Hz	20...20000 ms	HI PASS sets the cut-off frequency of the Hi pass filter.
LO PASS 2000 Hz	2000 Hz	20...20000 ms	LO PASS sets the cut-off frequency of the Lo pass filter..
STOP RIPPLE 40 dB	40 dB	21...100 dB	STOP RIPPLE sets the slope of the FIR filter.
Filter Type Low Pass High Pass Band Pass			Allows selecting the FIR filter type of the corresponding output channel.

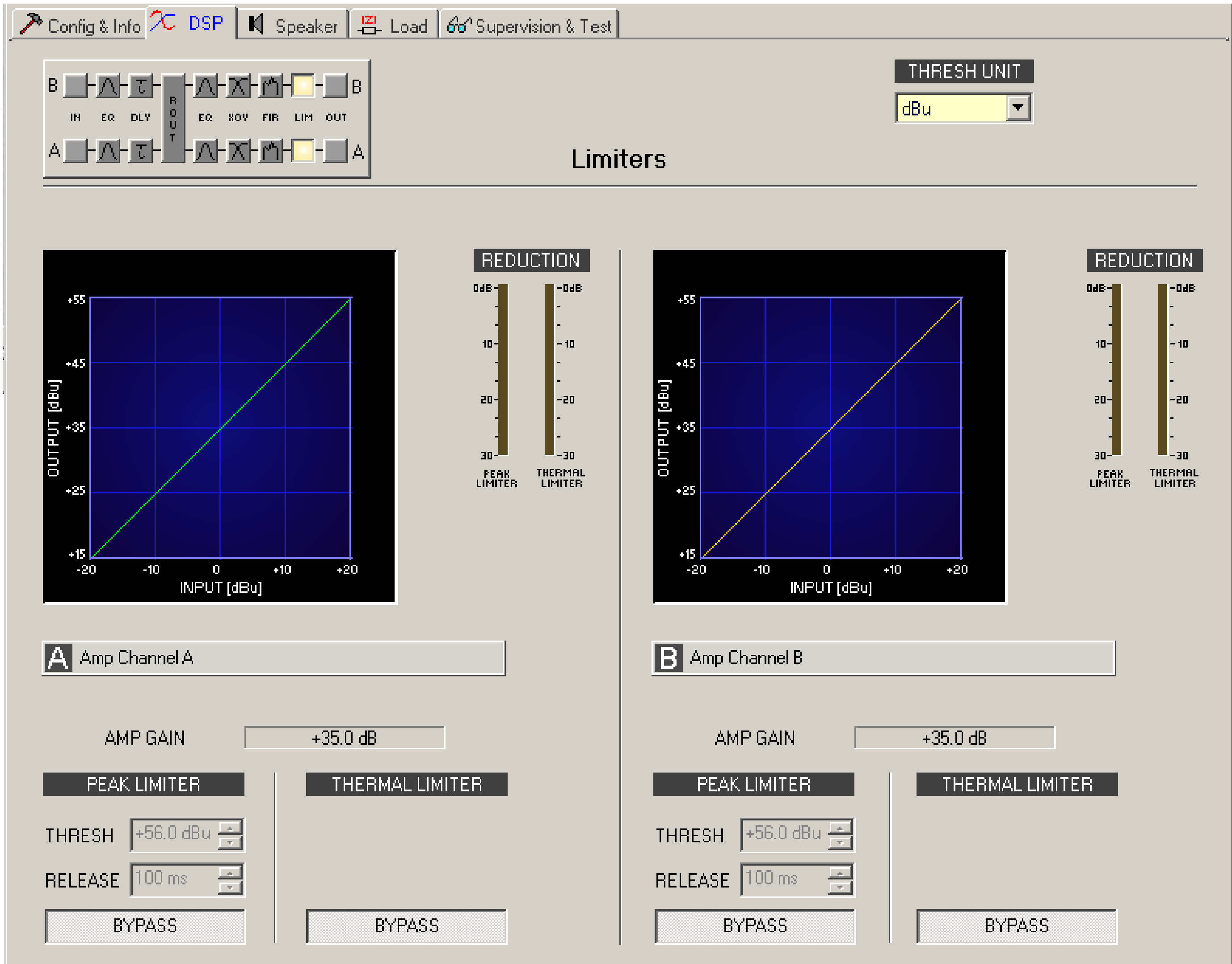
Filter Editing via “Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right.

LIMITERS

Each output channel of a remote amplifier offers a Peak Anticipation limiter and a Thermal limiter. These functions can be accessed via the Limiters window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and (thermal) overload.

Clicking onto the eight block in the Flow Diagram Selector or double clicking onto the LIMITERS block in the large flow diagram opens the Limiters window.



Channel Parameters

Element	Description
	Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all limiters parameters of the corresponding channel to any other channels within the same project.

Peak Limiter Parameters

Element	Default	Range	Description
PEAK LIMITER			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all peak limiter parameters of the corresponding channel to any other channels within the same project.
THRESH +55.7 dBu	+56 dBu	+26.0...+56.0 dBu or 15.46...488.99 V	THRESHOLD determines the audio signal level above which the limiter starts operating.
RELEASE 250 ms	100 ms	10...999 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
BYPASS			BYPASS switches the peak limiter on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the limited and non-limited audio signals.

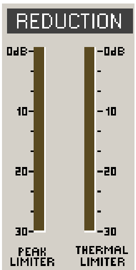
Thermal Limiter Parameters

Element	Default	Range	Description
THERMAL LIMITER			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all thermal limiter parameters of the corresponding channel to any other channels within the same project.
BYPASS			BYPASS switches the thermal limiter on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the limited and non-limited audio signals.

General Parameters

Element	Default	Range	Description
THRESH UNIT dBu	dBu	dBu / Volts	This lets you select the unit for the threshold parameter.

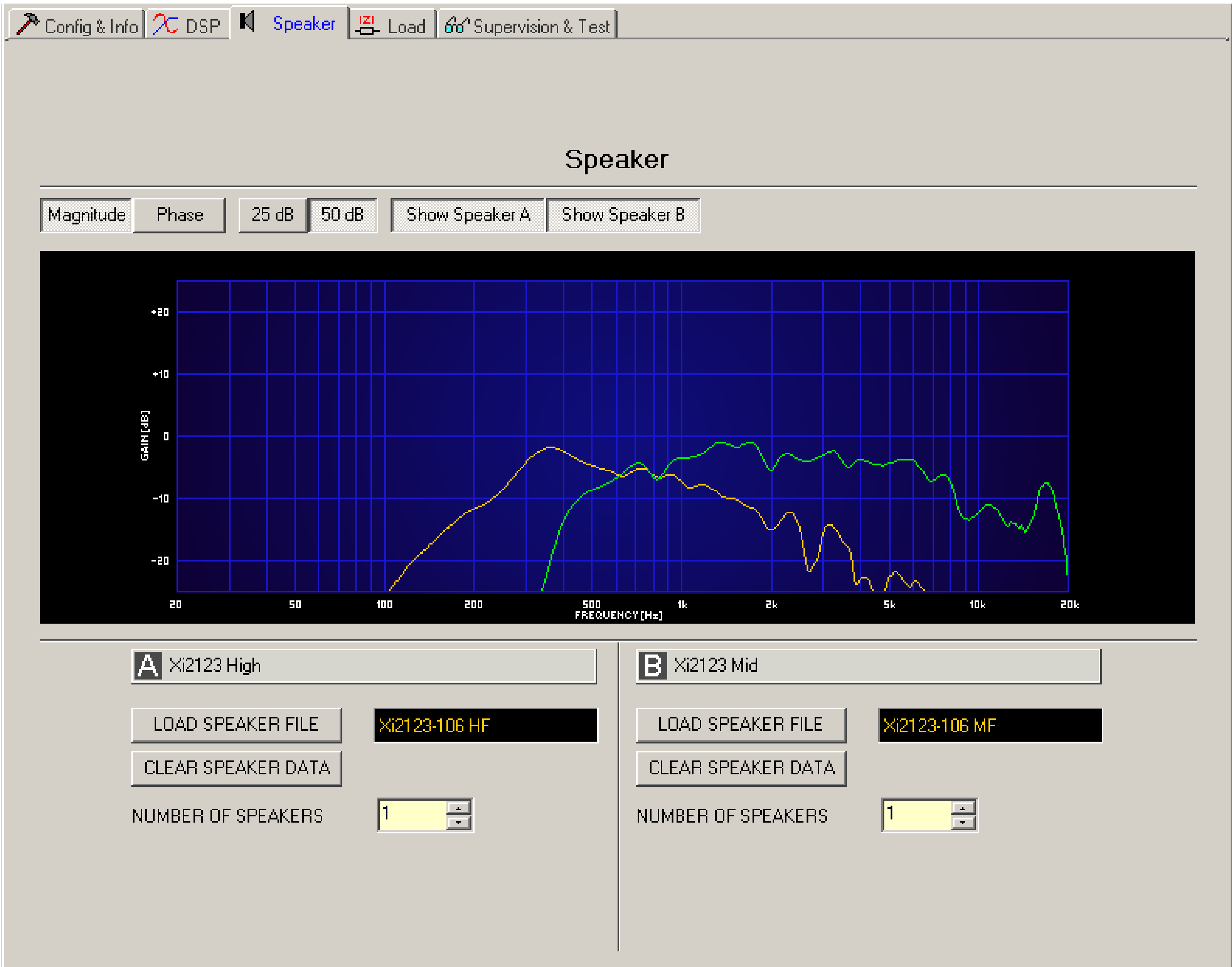
Indications

Element	Description
	These indicators show the reduction in dB that is applied to the audio signal by the peak limiter or thermal limiter. Level reduction is indicated as vertical yellow bar graph.

Editing Limiter Parameters by Dragging the Mouse in the Graphics Display
Active limiters (bypass button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding limiter by vertically dragging the mouse.
A limiter’s title lights in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

Speaker

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the amplifier channels and display the acoustic results of this virtual combination. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of all common Electro-Voice loudspeaker systems. Some examples are provided in the IRIS-Net directory Speaker Files.
HINT: When importing a speaker setting into a output channel the corresponding speaker file is imported automatically.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the amps. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of amplifier plus speaker transfer functions.
Clicking on the Speaker tab in the Setup & Control window opens the Speaker page.



Indication on the Graphic Display

Element	Description
Magnitude Phase	Switch for toggling between frequency response (magnitude) and phase response (phase) display
25 dB 50 dB	Switch for adjusting the scale of the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Speaker A Show Speaker B	Switching the display of the corresponding speaker data for an amplifier channel on/off is performed using the ”Show Speaker A” and ”Show Speaker B” switches.

Channel Parameters

Element	Default	Range	Description
A Lo (Sb121)			Channel description and channel name

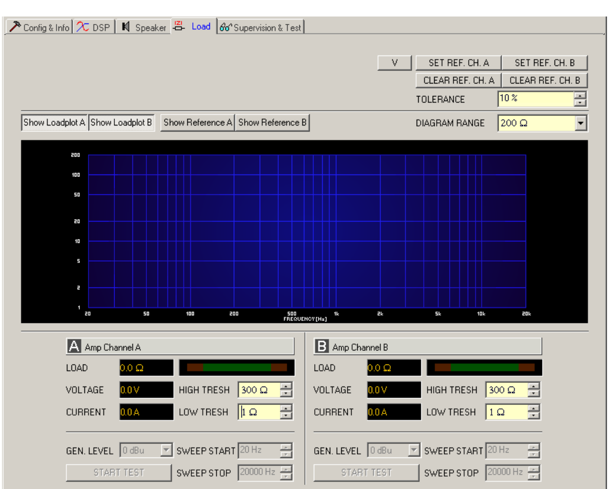
LOAD SPEAKER FILE Sb-121			Clicking the button LOAD SPEAKER FILE opens a dialog that allows the selection of the desired speaker file. Different loudspeaker files can be found in the Speaker files IRIS-Net folder. The name of the loaded loudspeaker model is shown in the black-shaded field on the right.
CLEAR SPEAKER DATA			Clicking the CLEAR SPEAKER DATA button clears the previously loaded measured speaker data of the selected channel.
NUMBER OF SPEAKERS 1	1	1...8	The NUMBER OF SPEAKERS parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel. Set- ting an amount from 1 to 8 is possible.

Load

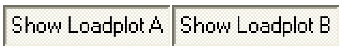
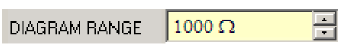
The Load window provides access to all settings and functions for testing and monitoring the load connected to the amplifier outputs.
The constantly measured output voltage and output current values of the Remote Power Amplifiers are indicated within the Load window. As soon as the output voltage of the signal present exceeds 150 mV, the resulting load is calculated and indicated. If the set thresholds are being exceeded or fallen short of, a corresponding message appears in the Load display of the Amplifier Control Panel. This dialog box permits to independently set the upper and lower thresholds for each power amp channel.
Within the Load window it is also possible to measure speaker impedance graphs and save them as references. The frequency range (start frequency, stop frequency) and the generator level of the sine-sweep test signal that is generated for this test can be adjusted. Specifying a tolerance field for the saved reference graphs is possible as well. A fault message is displayed in the event that a measurement exceeds or falls short of the tolerance range during system check.

HINT: The speaker impedance test is optimized for low impedance.

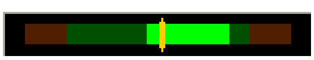
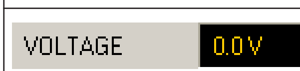
Select the Load window by clicking on the Load tab in the Setup & Control Window.



Graphic Display Indication

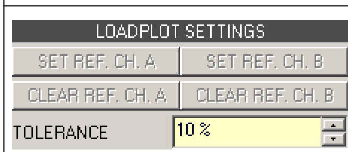
Element	Default	Range	Description
			The switches "Show Load plot A" and "Show Load plot B" turn the indication of the corresponding impedance graphs ON or OFF.
			The switches "Show Reference A" and "Show Reference B" turn the indication of the corresponding reference impedance graphs ON or OFF.
			The switch V toggles the unit of the Y-axis between Ohm and Volt.
	1000 Ohm	50 Ohm...10 kOhm or 0.5 V... 100 V	DIAGRAM RANGE allows zooming in or out the diagram's impedance range (Y- axis).

Parameters and Indications for the Continuous Monitoring of the Load Connected

Element	Default	Range	Description
			The load display indicates the quotient of measured voltage and current (U/I).
			This indication shows the actual measured load, the progression, and the set value range. The orange needle indicates the actual value. The bright green bar indicates which loads have already been measured while being on-line. A red indication signals that the value exceeded or fell short of the set value range. The dark green area represents the allowable value range for the load of the corresponding power amp channel. The set HIGH THRESH respectively LOW THRESH values define the limits for this value range. Moving the cursor over the indication bar brings up a tool-tip context menu showing the numerical value of the lowest, the highest, and the actually measured load values. Clicking with the right mouse button on the indication bar, followed by a click on Reset, clears the previously measured load values (bright green and red ranges disappear).
			The VOLTAGE display provides continuous indication of the corresponding power amp channel's output voltage.
			The CURRENT display provides continuous indication of the corresponding power amp channel's output current.

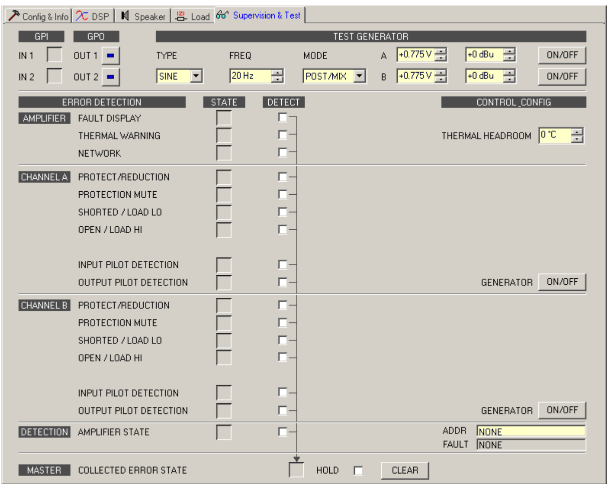
	300 Ohm	0.0 Ohm...70 kOhm	HIGH THRESH sets the upper limit of the allowable impedance range (= minimum load). Once this value is exceeded, an OPEN fault message (line interrupt) appears in the Amplifier Control Panel.
	1.0 0hm	0.0 Ohm...70 kOhm	LOW THRESH sets the lower limit of the allowable impedance range (= maximum load). Once this value is fallen short of, a SHORTED fault message (line short-circuit) appears in the Amplifier Control Panel.

Parameters For Impedance Measurement

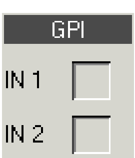
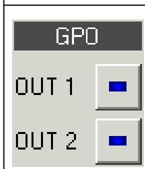
Element	Default	Range	Description
	-40 dB	-45...-5 dB	GEN. LEVEL sets the generator level for speaker impedance testing. **CAUTION: Extremely high levels during measurement may result in seriously damaging connected components.**
	20 Hz	20 Hz...20 kHz	SWEEP START sets the start frequency of the sine-sweep signal for speaker impedance testing.
	20 kHz	20 Hz...20 kHz	SWEEP STOP sets the stop frequency of the sine-sweep signal for speaker impedance testing.
			Clicking the START TEST soft-key launches the speaker impedance test. The generated sine-sweep signal sweeps over the previously defined frequency range. The graph of the measured impedance values is indicated in the Load plot display. Clicking this soft-key again cancels the test at any time.
	10%	5%...50%	Clicking the SET REF. CH. A and/or SET REF. CH. B soft-key saves the last test as reference. Clicking the CLEAR REF. CH. A and/or CLEAR REF. CH. B soft-key clears the corresponding reference. TOLERANCE defines the allowable deviation of the impedance graph. Actual measured test results and saved tolerance ranges are being compared during system check. A fault message is displayed if any point of the actual measurement falls outside of this tolerance range. The tolerance range is graphically displayed as spread in the corresponding color that surrounds the reference graph.

Supervision & Test

The Supervision & Test Dialog integrates functions for testing and monitoring power amps.
You can check control input states and trigger control outputs. A testing generator that provides sine, pink noise and white noise signal output allows acoustical testing. Status indicators for general power amp operation, the two amplifier channels and the load connected, indicate whether everything is okay or where failures occurred. You have the option to choose, which errors are combined and indicated in a general fault message.
A click on the Supervision & Test tab selects the page while in the Setup & Control Window.



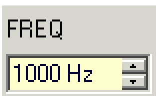
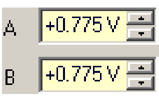
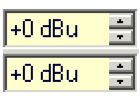
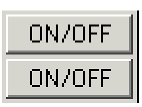
CONTROL INPUTS AND CONTROL OUTPUTS

Element	Description
	This dialog indicates the actual states of the two freely programmable control inputs IN1 and IN2. A green indicator signals "not active", i.e. the control input is open or "high". A red indicator signals "active", in that case the control input is connected to the ground or "low".
	This dialog is for manually controlling the two Open Collector control outputs OUT1 and OUT2. Not engaged (blue) indicates that the control output is deactivated or highly resistive while engaged (red) indicates that the control output is activated and connected to the ground (closed). **HINT: When a control output has already been programmed, the programmed function defines the state of the control output and manual control is not possible.**

For detailed explanation on how to program control inputs and outputs please refer to chapter Config & Info.

TEST GENERATOR PARAMETERS

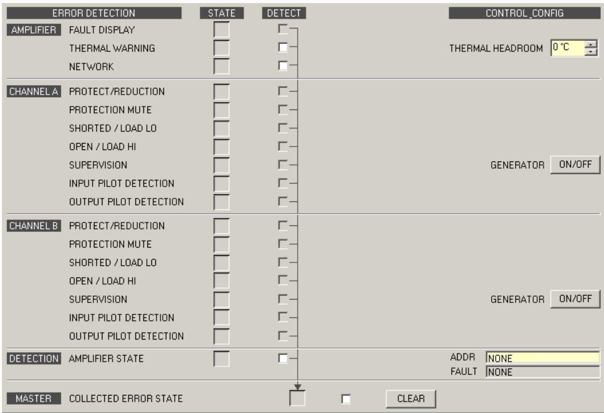
The test generator allows outputting a selected test-tone at an adjustable level via the power amps channel A and/or channel B which allows testing the cable run from the amplifier output to the connected loudspeaker systems as well as testing the functionality of the loudspeaker components

Element	Default	Range	Description
	SINE	SINE, WHITE, PINK	Type selects the test tone's signal-type. Available choices are: sine signal, white noise or pink noise.
	1000 Hz	20...20000 Hz	Freq defines the frequency of the sine signal. This parameter is not available when WHITE or PINK has been chosen as a test-tone signal.
	PRE/SOLO	POST/MIX, POST/SOLO, PRE/MIX, PRE/SOLO	MIX/SOLO determines whether the generated signal should be mixed with an existing signal. PRE/POST determines if the signal should be generated at the front (PRE) or the rear (POST) of the signal processing chain.
	0.775 V	0.001...2.451 V	These controls are for setting output voltage [V] of the corresponding amplifier outputs.
	0 dBu	-60...+10 dBu	These controls are for setting output level [dBu] of the corresponding amplifier outputs.
	OFF	OFF, ON	These ON/OFF push-buttons activate or deactivate test-tone signal output via corresponding amplifier channels. **CAUTION: Make sure to set a suitable output level, before activating the generator. Extreme output levels can lead to permanent damage of the connected loudspeaker systems!**

Error Detection

Error detection lists the individual STATE of fault indications. Errors collected are amp failure, channel failure, cable interruption, short-circuits, load deviation, ground fault, erroneous communication via the CAN bus as well as fault messages of other amps. A green STATE indicator signals normal operation. A red STATE indicator signals error detection.

If one of the corresponding DETECT boxes is marked, the state of that message is additionally included in the COLLECTED ERROR STATE. When activating the HOLD option, the indicator stays red after the occurrence of an error. If the HOLD option is not active, indication returns to green, once the fault is not detected anymore. Pressing the CLEAR button in the COLLECTED ERROR STATE line resets the indicator from red to green and stored errors are deleted. The COLLECTED ERROR STATE indicator resembles exactly the Amplifier State indicator of the System Check Window. The collected fault state message can be outputted via a control output. For detailed explanation please refer to chapter Config & Info.



Error art	Description
FAULT DISPLAY	The red FAULT DISPLAY indicator lights, if a error message is indicated at the amplifier's front LC-Display.
THERMAL WARNING	The power amplifier is protected against thermal overload and will reduce the output power if the internal temperature exceeds a fixed threshold value (please refer to the owner‘s manual for more details). In this case the THERMAL WARNING indicator lights red. With the THERMAL HEADROOM parameter in IRIS-Net the threshold for the THERMAL WARNING indication can be changed to get the warning indication already at lower temperatures, meaning before the amplifier reduces the output power. The THERMAL HEADROOM can be adjusted from 0 to 30 °C which means a temperature warning indication could be configured up to 30 °C before reduction.
NETWORK	This indicator shows whether communication at the CAN bus interface is normal (green) or when a problem exists (red). The power amp automatically detects whether commands from a PC or another central control unit are missing and signals the problem via the Communication Flag.
PROTECT / REDUCTION	When the red PROTECT / REDUCTION indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated. The differentiated protections concept of the power amp results in several protection circuits being activated one after another, which ensures that under normal circumstances the power amplifier will stay in the safe and stabile opera- ting range.
PROTECTION MUTE	When the red PROTECT MUTE indicator lights, the signal path gets switched off. The amplifier needs to be switched off to prevent power amplifier and connected speaker systems from being damaged, this is indicated by the PROTECT and MUTE-LEDs being lit simultaneously.
SHORTED / LOAD LO	This indicator lights red when the measured impedance value of the corresponding power amp output falls below a pre-set minimum or when it is shorted. Setting the minimum value is possible in the Load dialog.
OPEN / LOAD HI	This indicator lights red when the measured impedance value of the corresponding power amp output exceeds a pre-set maximum or when cable interruption is detected. Setting the maximum value is possible in the Load dialog.

INPUT PILOT DETECTION	A remote amplifier's audio inputs support pilot-tone detection and evaluation. Using an externally generated pilot-tone signal allows the monitoring of audio cables and analog input stages. The threshold for 19 kHz pilot-tone evaluation is set to -40 dBu / 7.75 mV. The indicator lights green when an external pilot-tone signal coming from mixer, matrix, controller, etc. is detected. A missing pilot-tone signal or a drop in its level below the evaluation threshold causes the indicator to change to red. Only mark the DETECT box next to the indicator when an external pilot-tone signal actually exists and input monitoring has been configured.
OUTPUT PILOT DETECTION	This indicator is for amplifier monitoring via external pilot-tone signal. In that case, internal pilot-tone generation needs to be switched off to avoid interference between the two signals. Detection and evaluation is performed at the amplifier output. The indicator lights green when a 19 kHz pilot-tone signal with a level of at least -14 dBu / 150mV is detected. A missing pilot-tone signal or a drop in its level below -14 dBu (threshold) results in error detection. The indicator changes to red. **CAUTION: The externally fed pilot-tone signal passes through the entire signal path of the remote amplifier, i.e. the signal is influenced by filtering and x-over settings. When setting the external pilot-tone generator's level, make sure to mind possible amplification/ attenuation applied by internal filters.**
AMPLIFIER STATE	A RCM-26 remote amplifier is capable of detecting and indicating the operational state of other RCM-26 amps within a CAN network. The addresses of all amps that are to be monitored are entered in the ADDR field, e.g. 2-4,6,11. The FAULT field indicates the amp addresses for which errors have been detected and the COLLECTED ERROR STATE has been activated (red). The indicator changes to red as soon as at least one amplifier in the list shows erroneous operation.
COLLECTED ERROR STATE	COLLECTED ERROR STATE is a collected fault message that combines all error types detected for which the DETECT box had been mar- ked. The HOLD function allows keeping the COLLECTED ERROR STATE for later evaluation while CLEAR clears the indication after remedying the cause of the fault. The COLLECTED ERROR STATE indication is identical to the indication in the Amplifier Status column within the RCM-26 System Check Window.

RS-232 Protocol for RCM-26

The RS-232 port is located on the rear panel of RCM-26 remote power amps. It can be used as interface for the connection of media control systems or facility management systems. RS-232 allows controlling and polling all parameters. Communication is performed using an easy to implement ASCII protocol which allows easy integration of remote amps in media and/or touch-panel applications. For programming notes and a complete description of the protocol please refer to the following chapters.

RS-232 SETTINGS

The RS-232 interface of the RCM-26 remote power amps is permanently configured allowing full duplex operation. Set values are:

Parameter	**Setting**
Baud Rate	19200 bits per second
Data Bits	8
Parity	None

Stop Bits	1
Flow Control	Xon / Xoff

The command string "*** RCM-26 command mode entered ***" is sent to RS-232 after powering on the remote amp and after a short initializing period. The RS-232 interface is now ready for communication.

ASCII CONTROL PROTOCOL

A simple ASCII string protocol, which is referred to as ASCII Control Protocol is implemented in the remote amps. Commands are organized in a tree structure with up to 6 levels. The slash "/" or a space " " can be used for separation. The question mark "?" can be utilized to query parameter settings or commands of the corresponding level. To step one level down you have to enter "../".

The following table lists the ASCII Control Protocol commands with brief explanations.

Level1	Level2	Level3	Level4	Level5	Read Write	Values	Description
							Commands for RS232 communication
/COMM	/LINEFEED				R / W	ON, OFF	Linefeed state for RS232 communication
	/PROMPT				R / W	ON, OFF	Prompt state for RS232 communication
	/ECHO				R / W	ON, OFF	Echo state for RS232 communication
							Amplifier / Channel Names
/NAME	/DEVICE				R / W	up to 30 characters	Amp name
	/IN_A				R / W		Input A name
	/IN_B				R / W		Input B name
	/OUT_A				R / W		Output A name
	/OUT_B				R / W		Output B name
							Amplifier Power On / Stand-by and Operational State
/POWER	/SWITCH				R / W	ON, OFF	Switch amp ON / OFF or read out ON / OFF state
	/DELAY				R / W	0...6.35 s	Power-On-Delay in seconds.
							Connect/Disconnect Amplifier
/ SERVICE	/CAN	/ CONNE CT				0...255	Transparent ASCII Control Protocol connection via CAN to remote RCM, write CAN address (1...250) of RCM to connect to, or write 0 to disconnect. Active remote connection is shown as address in brackets before the prompt.

							Commands for Level Indication
/METER	/READ				R		Read out all actual vu and output U/I values
							Commands for Amplifier Temperature Indication
/TEMP	/MODULE				R	10...137.5 [°C]	Read out actual RCM-26 temperature
	/SUPPLY				R	10...137.5 [°C]	Read out actual power supply temperature (primary side)
	/ATMO				R	10...137.5 [°C]	Read out actual atmosphere temperature (in chassis)
	/CHAN_A	/AMP			R	10...137.5 [°C]	Read out actual channel A output amplifier temperature
		/ SUPPLY			R	10...137.5 [°C]	Read out actual power supply temperature (channel A, secondary side)
		/TH_HR			R	0...127.5 [K]	Read out thermal headroom of channel A
		/ TH_TRS	/TRS		R / W	20...150 [°C]	Threshold for thermal headroom (THRM_HDRM) flag. The flag is set as soon as the headroom falls below this threshold.
		/ TH_TRS	/HYS		R / W	0...40 [°C]	Hysteresis for thermal headroom (THRM_HDRM) flag. The flag is deleted, as soon as the headroom rise above the threshold plus hysteresis.
	/CHAN_B	...			R		Same as above for input channel B
	/OVT_TRS	/TRS			R / W	20...150 [°C]	Threshold for thermal overload (OVT) flag. The flag is set as soon as the temperature threshold is reached by at least one temperature value.
		/HYS			R / W	0...40 [°C]	Hysteresis for thermal overload (OVT) flag. The flag is deleted, as soon as the temperature falls below the threshold minus hysteresis.
							Commands for Audio Monitoring

/MONI					R / W	NONE, RELAY, IN_A, OUT_A, IN_B, OUT_B	List of active elements for Audio Monitoring. Input and output channels can be monitored. RELAY switches active channels onto the monitor bus.
							Commands for DSP Parameters
/PRM	/IN_A	/IDX00			R / W		Read and write of input channel A DSP parameter values via index numbers. For further details please refer to DSP Parameter Index Table
		...					
		/IDX2B			R / W		
	/IN_B	...					Same as above for input channel B
	/OUT_A	/IDX00			R / W		Read and write of output channel A DSP parameter values via index numbers. For further details please refer to DSP Parameter Index Table
		...					
		/IDX41			R / W		
	/OUT_B						Save as above for output channel B
	/DLYTEMP				R / W	-20.0...+60.0 [°C]	Ambient temperature for the calculation of delays with distance values.
							Commands for Presets
/PRESET	/LOAD				R / W	U1...U6, O1...O2, F1...F2	Load User Presets U1...U6, Owner Presets O1...O2 or Factory Presets F1...F2. Readout of preset data loaded last. The string "(edited)" behind the preset number indicates that values have already been edited.
	/SAVE				W	U1...U6, O1...O2	Save User Preset or Owner Preset
	/TITLE				R / W	up to 16 characters	Preset Name
	/INITIAL				R / W	U1...U6, O1...O2, F1...F2, A	Selection of user/owner/factory/actual preset to be loaded after reset
	/PRSGATE				R / W	ON, OFF	State of identical named error/statusflag which pre- vents from dangerous changes of output configuration.
	/OWNER	/ PASSWORD			R / W	OK, FALSE, "password"	Write "password" to identify owner until next reset. Saving owner presets and password definition can only be done if this state is 'OK'. Read value (OK, FALSE) if the owner is identified successfully.

		/PROTECT			R / W	ON, OFF, "new Password"	Read if an owner password is already defined (ON, OFF). Write "new Password" to define a new password (up to 16 characters). Writing is only possible if /PRE- SET/OWNER/ PASSWORD is "OK".
							Commands for Control Inputs/Control Outputs
/CONTROL	/IN1	/STATE			R	ON, OFF	State of the control input
		/ON	/TIME		R / W	0...10.0 [s]	Delay / debounce time during activation
			/FNCT		R / W	NOTHING, POWER, ABS, REL, TOG- GLE, PRESET, MONI, MEMFLAG, MEAS, TEST- GEN	Function during activation. For further details please refer to the table "Control Inputs - GPI Functions" below.
			/PRM	/...			Parameter and values for the functions mentioned before
		/OFF	...				(same as above but for the deactivation of control inputs)
	/IN2	...					(same as above but for the control input 2)
	/OUT1	/STATE			R / W	ON, OFF	State of the Control Output
		/ON	/TIME		R / W	0...10.0 [s]	Delay / debounce time for the programmed condition
			/FNCT		R / W	NOTHING, POWER, ABS, TEMP, VU, CTL_IN, ERR- FLAG, MEMFLAG, PRESET	Condition that activates a control output. For further details please refer to the table "Control Outputs - GPO Functions" on this page.
			/INV		R / W	ON, OFF	Inverts the result of the programmed condition

			/SYNC		R / W	ON, OFF	Lets you select whether the control outputs can be synchronized using a special CAN-command.
			/PRM	/..			Parameters and values for functions mentioned above
		/OFF	...				(same as above but for switching off a control output)
	/OUT2	...					(same as above but for the control output 2)
	/MEMFLAG	/SET			R / W	NONE, 1...16	List of currently set Memo flags
		/CLR			R / W	NONE, 1...16	List of currently reset Memo flags
							Commands for Receive and Transmit job codes
/JOB	/RX1	/ID			R / W	0...FFFF	Number (ID) in hex for job code to be received. Each power amp can receive and interpret up to 5 job codes.
		/FNCT			R / W	NOTHING, POWER, ABS, REL, TOG- GLE, PRESET, MONI, MEMFLAG, MEAS, TEST- GEN	Function when receiving a job code. For further details please refer to table "Job Codes - Receive Functions" on this page.
		/PRM	/...				Parameters and values for functions mentioned above
	...						
	/RX5	...					(same as above but for receiving Job Codes 2 to 5)
	/TX1	/ID			R / W	0...FFFF	Number (ID) for job code to be transmitted. Each power amp can transmit up to 5 job codes.
		/TIME			R / W	0...10.0 [s]	Delay / debounce time for programmed condition

		/FNCT			R / W	NOTHING, POWER, ABS, TEMP, VU, CTL_IN, ERR- FLAG, MEMFLAG, PRESET	Condition that triggers the transmission of a job code. For further details please refer to table "Job Codes - Transmit Functions" on this page.
		/INV			R / W	ON, OFF	Inverts the result of the programmed condition
		/PRM	/...				Parameters and values for functions mentioned above
	...						(same as above but for transmitting Job Codes 2 to 5)
	/TX5	...					
	/LAST	/RX			R / W	0000...03FF	The ID (hex code) of the last received job code is displayed during reading. Writing simulates the reception of a job code with the stated ID (hex code) by the power amp.
		/TX			R / W	0000...03FF	The ID (hex code) of the last transmitted job code is displayed during reading. Writing transmits a job code with the stated ID (hex code).
							Commands for Error and Status Requests
/STFLAG	/DEVICE	/ACT			R / W	see table Device State Flags	List of currently set status and error flags of device. Writing resets some flags.
		/ COLLE CT			R / W		Flag template for Collected Error Flag (a list of status and error flags as mentioned above). The state is buffered (Hold function) when COLLECT is listed in the template.
	/IN_A	/ACT			R / W	see table Input Channel Flags	List of currently set status and error flags of input A. Writing resets some flags.
		/ COLLE CT			R / W		Flag template for Collected Error Flag (a list of status and error flags as mentioned above). The state is buffered (Hold function) when COLLECT is listed in the template.

	...						
	/IN_B	...					(same as above for input channel B)
	/OUT_A	/ACT			R / W	see table Out- put Channel State flags	List of currently set status and error flags of output A. Writing resets some flags.
		/ COLLE CT			R / W		Flag template for Collected Error Flag (a list of status and error flags as mentioned above). The state is buffered (Hold function) when COLLECT is listed in the template.
	...						
	/OUT_B	...					(same as above for output channel B)
							Commands for analog and digital (AES/ EBU) audio inputs
/INPUT	/MANUAL				R / W	AD, AES	Read out actual used audio input. Writing selects the analog or digital audio input.
	/AUTO	/ ACTUAL			R / W	AD, AES	Read out actual used audio input. Writing selects the analog or digital audio input.
		/FB			R / W	ON, OFF	Activates the automatic fallback from digital to analog audio input.
		/ FB_TIM E			R / W	0...1000.0 [s]	Automatic fallback time in seconds. With automatic audio input selection enabled and AES audio input status not ok for longer than this time, audio input selection will fall back from AES to analog input.
		/FF			R / W		Activates the automatic fall-forward from analog to AES input
		/ FF_TIM E			R / W		Automatic fall-forward time in seconds. With automatic audio input selection enabled and AES audio input status ok for longer than this time, audio input selection will fall forward from analog to AES input.
	/AES	/LOCK			R	ON, OFF	Read out actual lock stat of AES input
		/SF			R		Read out sample rate of AES input signal

Examples:

- /POWER/SWITCH ON ; switches amp's power on
- /TEMP/MODULE ? ; queries the remote control module temperature
- /TEMP/MODULE 65 ; reply to query: 65 °C
- /STFLAG/DEVICE/ACT ? ; queries operational state and error flags
- /STFLAG/ACT POWER,GLOBAL ; reply to query: Power is On, Global Error detected (Collected Error in external CAN-devices)
- /STFLAG/GLOBAL ? ; queries for which external CAN-devices errors have been detected
- /STFLAG/GLOBAL 3-4 ; reply to query: Collected Error Flags on amps 3 and 4 are set

DSP PARAMETER INDEX TABLE
Input channel A or B

Index	Parameter Description	Value Range	Value Description
/IDX00	delay bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX01	delay	1.68467...2000	ms
/IDX02	eq1 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX03	eq1 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX04	eq1 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX05	eq1frequency	20...20000	Hz
/IDX06	eq1 gain	-18...12	dB
/IDX07	eq1 quality	0.4...40	
/IDX08	eq2 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX09	eq2 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX0A	eq2 slope	0...2	0=0dB bypass, 1=6dB, 2=12dB
/IDX0B	eq2frequency	20...20000	Hz
/IDX0C	eq2 gain	-18...12	dB
/IDX0D	eq2 quality	0.4...40	
/IDX0E	eq3 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX0F	eq3 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX10	eq3 slope	0...2	0=0dB bypass, 1=6dB, 2=12dB
/IDX11	eq3frequency	20...20000	Hz
/IDX12	eq3 gain	-18...12	dB
/IDX13	eq3 quality	0.4...40	
/IDX14	eq4 bypass	0 / 1	0 = ON, 1 = BYPASS

/IDX15	eq4 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX16	eq4 slope	0...2	0=0dB bypass, 1=6dB, 2=12dB
/IDX17	eq4frequency	20...20000	Hz
/IDX18	eq4 gain	-18...12	dB
/IDX19	eq4 quality	0.4...40	
/IDX1A	eq5 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX1B	eq5 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX1C	eq5 slope	0...2	0=0dB bypass, 1=6dB, 2=12dB
/IDX1D	eq5frequency	20...20000	Hz
/IDX1E	eq5 gain	-18...12	dB
/IDX1F	eq5 quality	0.4...40	
/IDX20	eq6 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX21	eq6 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX22	eq6 slope	0...2	0=0dB bypass, 1=6dB, 2=12dB
/IDX23	eq6frequency	20...20000	Hz
/IDX24	eq6 gain	-18...12	dB
/IDX25	eq6 quality	0.4...40	
/IDX26	testgenerator enable	0 / 1	0 = OFF, 1 = ON
/IDX27	testgenerator level	-128...+40	dB(u) at amplifier output, -128dB for Mute
/IDX28	not used		
/IDX29	not used		
/IDX2A	route	0...2	0=A, 1=B, 2=A+B
/IDX2B	19kHz pilot notch filter	0 / 1	0=disable, 1=enable

Output channel A or B

Index	Parameter Description	Value Range	Value Description
/IDX00	level	-128...+6	dB
/IDX01	mute	0 / 1	0 = ON, 1 = MUTE
/IDX02	route	0...3	0 = nothing, 1 = A, 2 = B, 3 = A+B
/IDX03	level trim	-30...+6	dB
/IDX04	delay bypass	0 / 1	0 = ON, 1 = BYPASS

/IDX05	delay	0...350	
/IDX06	polarity	0 / 1	0 = normal, 1 = inverted
/IDX07	compressor bypass	0 / 1	0 = ON, 1 = BYPASS (obsolete with firmware V1.15)
/IDX08	compressor type (ratio)	0...4	0 = 1/1, 1 = 1/1.4, 2 = 1/2, 3 = 1/4, 4 = 1/8 (obsolete with firmware V1.15)
/IDX09	compressor threshold	-30...0	dB (obsolete with firmware V1.15)
/IDX0A	compressor attack	0...99	ms (obsolete with firmware V1.15)
/IDX0B	compressor release	10...999	ms (obsolete with firmware V1.15)
/IDX0C	limiter bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX0D	limiter threshold	-30...0	dB
/IDX0E	limiter release	10...999	ms
/IDX0F	hipass xover type	0...17	0 = off, 1 = butter6, 2 = s12q05, 3 = s12q06, 4 = s12q07, 5 = s12q08, 6 = s12q10, 7 = s12q12, 8 = s12q15, 9 = s12q20, 10 = bessel12, 11 = butter12, 12 = linkwz12, 13 = bessel18, 14 = butter18, 15 = bessel24, 16 = butter24, 17 = linkwz24
/IDX10	hipass xover frequency	20...20000	Hz
/IDX11	lopass xover type	0...17	0 = off, 1 = butter6, 2 = s12q05, 3 = s12q06, 4 = s12q07, 5 = s12q08, 6 = s12q10, 7 = s12q12, 8 = s12q15, 9 = s12q20, 10 = bessel12, 11 = butter12, 12 = linkwz12, 13 = bessel18, 14 = butter18, 15 = bessel24, 16 = butter24, 17 = linkwz24
/IDX12	lopass xover frequency	20...20000	Hz
/IDX13	eq1 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX14	eq1 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX15	eq1 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX16	eq1 frequency	20...20000	Hz
/IDX17	eq1 gain	-18...12	dB
/IDX18	eq1 quality	0.4...40	
/IDX19	eq2 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX1A	eq2 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX1B	eq2 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX1C	eq2 frequency	20...20000	Hz
/IDX1D	eq2 gain	-18...12	dB

/IDX1E	eq2 quality	0.4...40	
/IDX1F	eq3 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX20	eq3 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX21	eq3 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX22	eq3 frequency	20...20000	Hz
/IDX23	eq3 gain	-18...12	dB
/IDX24	eq3 quality	0.4...40	
/IDX25	eq4 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX26	eq4 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX27	eq4 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX28	eq4 frequency	20...20000	Hz
/IDX29	eq4 gain	-18...12	dB
/IDX2A	eq4 quality	0.4...40	
/IDX2B	eq5 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX2C	eq5 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX2D	eq5 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX2E	eq5 frequency	20...20000	Hz
/IDX2F	eq5 gain	-18...12	dB
/IDX30	eq5 quality	0.4...40	
/IDX31	eq6 bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX32	eq6 type	0...5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX33	eq6 slope	0...2	0 = 0dB bypass, 1 = 6dB, 2 = 12dB
/IDX34	eq6 frequency	20...20000	Hz
/IDX35	eq6 gain	-18...12	dB
/IDX36	eq6 quality	0.4...40	
/IDX37	fir filter		can't be represented by numerical value
/IDX38	fir filter bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX39	testgenerator enable	0 / 1	0 = OFF, 1 = ON
/IDX3A	testgenerator level	-128...+40	dB(u) at power amp output
/IDX3B	not used		
/IDX3C	not used		
/IDX3D	pilotgenerator enable	0 / 1	0 = off, 1 = on

/IDX3E	pilotgenerator level	-128...+40	dB(u) at power amp output
/IDX3F	config type	0...7	0 = fullrange, 1 = sub, 2 = sublo, 3 = lo, 4 = lomid, 5 = mid, 6 = midhi, 7 = hi
/IDX40	config description		can't be represented by numerical value
/IDX41	speaker protection		can't be represented by numerical value
/IDX42	thermo limiter		can't be represented by numerical value

STATE/ERROR FLAGS

The available state/error flags are divided into following groups:

- Global power amplifier / remote control module (Device State Flags),
- Amplifier's input channels (Input Channel State Flags)
- Amplifier's output channels (Output Channel State Flags)

Device State Flags

Flag	Description
POWER	The amplifier is in powered state (after power up delay gone) (collect / GPO reads inverted)
STANDBY	The amplifier is in standby state
CANPOLL	There is a timeout at master PCs CAN polling
NONVLT	There is a error in non volatile memory handling
COLLECT	Locally collected error state detection via individual mask for device and all input/output channels (can be cleared by command)
GLOBAL	Globally collected error state of all selected other amps
OVT	Amplifier temperature is above upper limit
DIRTY	Current preset parameter setting is edited
PRSGATE	The possibility of a "dangerous" preset change is enabled
BRIDGED	The amplifier's bridged mode is selected.
PARALLEL	The amplifier's parallel routing is selected
GNDLIFT	The connection switch "chassis to ground" is open
LEVELCTRL	The level controls at the power amp's front panel are disabled
VOLTDOUB	The mains voltage-doubler is active, 115V-mode
THERMSD	The thermal shutdown is active
CBPWARN	The circuit-breaker-protection is close before action
AESLOCK	The AES/EBU digital audio interface is in locked state (collect / GPO reads inverted)
ERRDISP	The LC-display indicates a error message
AESERROR	The AES/EBU digital audio interface is in erratic state
VLT_WARN	Mains voltage warning

OWNER_PW	A password for owner presets protection is defined
OWNER_ID	The owner is identified by password, manipulation of owner presets is possible (can be cleared by command)
AES_OK	Die AES/EBU Schnittstelle ist in Ordnung
AESINPUT	Es wird das Audio-Signal der AES/EBU Schnittstelle statt dem Audio-Signal am analogen Eingang verwendet
CANADDR	Die CAN-Adresse wurde per Software gewählt, die Adress-Einstellung an der Blende des RCM-26 wird ignoriert

Input Channel State Flags

Flag	**Description**
PILOT	The 19 kHz pilot tone is not detected above threshold

Output Channel State Flags

Flag	**Description**
PSOVL	The SMPS reports overload
HFDET	The power amplifier reports excessive HF level
SHORTED	A low impedance load or short-circuit is detected
THERMPROT	The amplifier's thermal protection is active (protect-mode)
PROTECT	In general: protect-mode active
AMPFAIL	A amplifier failure was detected (output is muted via relay off)
MUTE	The output is muted via relay off
GNDFLT	A ground fault has occurred
GAINRED	The amplifier's gain reduction is active (protect-mode)
DISHI	Rail voltage: step high disabled
DISMID	Rail voltage: step mid disabled
Z_VALID	The calculated load value (U/I) is actually valid
Z_MIN	The load is below minimum, shorted
Z_MAX	The load is above maximum, open
PILOT	The 19 kHz pilot tone is not detected above threshold
MEAS	The sweep measurement is active
THRM_HDRM	The thermal headroom is below minimum

Control Inputs - GPI Functions

Every control input can be programmed with individual functions for switching on (/CONTROL/INx/ON/...) and switching off (/CONTROL/INx/OFF/...). When the state of a control input changes, the programmed function is executed after the previously set delay or debounce times are expired (up to 10 sec.). Available functions are explained in the following table.

Job Codes - Receive Functions

Job codes are distributed throughout the CAN network via broadcast commands. Each job code has a freely definable number (ID). Received job codes can trigger the same functions as local GPI control inputs. Receiving a job code with the defined number (ID) triggers the function with its specified parameter values. Available functions for /JOB/RXx/FNCT/... and corresponding parameters /JOB/RXx/PRM/... are identical with local GPI functions, as outlined in the table.

Function	**Parameter**	**Range**	**Description**
NOTHING			No function
POWER			Controls Power On / Stand-by
	../PRM/SWITCH	ON	Switches the amp's power to ON
		OFF	Switches the amp in Stand-by mode
		FLIP	Toggles between ON and Stand-by and vice versa
ABS			Sets the selected DSP parameter to an absolute value
	../PRM/TYPE	IN / OUT	IN selects input channel, OUT selects output channel
	.../PRM/CHAN	A / B / A,B	Selects channel A, or channel B, or channel A and B
	../PRM/IDX	00...FF	Selects the DSP parameter via index number
	../PRM/VALUE		New absolute parameter value
REL			Changes the selected DSP parameter in relation to the actual value
	../PRM/TYPE	IN / OUT	IN selects input channel, OUT selects output channel
	.../PRM/CHAN	A / B / A,B	Selects channel A, or channel B, or channel A and B
	../PRM/IDX	00...FF	Selects the DSP parameter via index number
	../PRM/VALUE		Relative change of the parameter
TOGGLE			Toggles a DSP parameter between 0 and 1 (this only makes sense for flag parameters, e.g. MUTE, BYPASS, etc.)
	.../PRM/TYPE	IN / OUT	IN selects input channel, OUT selects output channel

	.../PRM/CHAN	A / B / A,B	Selects channel A, or channel B, or channel A and B
	../PRM/IDX	00...FF	Selects the DSP parameter via index number
PRESET			Loads a DSP preset
	../PRM/NR	U1...U6, F1...F2, O1...O2	Selects an user preset User Preset (Ux), a Factory Preset (Fx) or a Owner Preset (Ox)
MONI			Controls the selection for the audio monitoring bus
	../PRM/SEL	NONE, RELAY, IN_A, OUT_A, IN_B, OUT_B	Selects audio monitoring parameters. All combinations are possible.
	../PRM/SWITCH	ON, OFF	Switches the selected audio monitoring parameter ON or OFF
MEMFLAG			Manipulates general Memo flags
	../PRM/CLR	NONE, 1...16	Clears selected flags
	../PRM/TOGGLE	NONE, 1...16	Changes the state of selected flags. Use CLR and TOGGLE together, so that selected flags are set afterwards.
MEAS			Initiates impedance testing at a fixed frequency
	../PRM/CHAN	A / B / A B	Selects channel A, or channel B, or channel A and B
	../PRM/FREQU	10...20000 [Hz]	Generator frequency for impedance test
	../PRM/LEVEL	-128...+50 [dBu]	Generator level for impedance test
	../PRM/TIME	0.0, 0.1...1000 [ms]	Impedance test time span. 0.0 = continuously ON
	../PRM/Q	0.0...60	Quality of band pass filter
	../PRM/MIX	ON, OFF	Wanted signal and Generator signal mixed
	../PRM/PRE	ON, OFF	Generator signal fed in at the input (ON) or output (OFF) of the DSP signal chain
TESTGEN			Defines parameters for the audio testing generator
	../PRM/INPCHAN	A / B / A,B	Selects input channel A, or input channel B, or input channel A and B

	../PRM/ OUTCHAN	A / B / A,B	Selects output channel A, or output channel B, or output channel A and B
	../PRM/LEVEL	-128...+50 [dBu]	Defines the testing generator output level
	../PRM/MODE	OFF, SINE, WHITE, PINK	Defines the testing generator's signal type
	../PRM/FREQU	10...20000 [Hz]	Defines the generator frequency, when SINE is selected
	../PRM/MIX	ON, OFF	Wanted signal and Testing generator signal mixed
AUDIO_IN			Selection of audio input
	.../PRM/SWITCH	AD, AES, FLIP	Select the analog audio input (AD), the AES/EBU input (AES) order toggles (FLIP) audio input selection from analog to AED/EBU and vice versa

CONTROL OUTPUTS - GPO FUNCTIONS

Two conditions can be programmed for each control output which either activate the output (/CONTROL/OUTx/ON/...) or deactivate the output (/CONTROL/OUTx/OFF/...). When the assigned function (/CONTROL/OUTx/ON/FNCT or / CONTROL/OUTx/OFF/FNCT) is recognized as "true" and the state is maintained for at least the set delay or debounce times (up to 10 sec.), the control output changes to activated (On) or deactivated (Off). The INV parameter allows inverting the state of the assigned function. Synchronizing the switching of control outputs is possible by means of a special system-wide CAN command, when SYNC is set to ON. Available functions and corresponding settings are explained in the following table.

JOB CODES - TRANSMIT FUNCTIONS

Job codes are distributed throughout the CAN network via broadcast command. Each job code has a freely definable number (ID). Identical conditions can be assigned to job codes and control outputs. A job code with a defined number (ID) is transmitted, when the corresponding condition for (/JOB/TXx/FNCT) is recognized as "true" and the state is maintained for at least the set delay or debounce times (up to 10 sec.). The INV parameter allows inverting the state of the assigned function. Available functions for /JOB/TXx/FNCT/... as well as corresponding parameters /JOB/TXx/ PRM/... are identical to local GPO functions, as outlined in the table.

Function	Parameter	Range	Description
NOTHING			No function
POWER			Interpretation results in "true", when the power amp is powered on (even during power-on delay) and "false", when the amp's power is off.
ABS			Interpretation results in "true", when the DSP parameter value is higher or equals the reference value.
	../PRM/TYPE	IN / OUT	IN selects input channel, OUT selects output channel
	../PRM/CHAN	A / B / A,B	Selects channel A, or channel B, or channel A and B
	../PRM/IDX	00...FF	Selects the DSP parameter via index number

	../PRM/VALUE		Reference value for comparison
TEMP			Interpretation results in "true", when the measured amplifier temperature is higher or equals the reference value.
	../PRM/ CELSIUS	-20...150 [°C]	Temperature reference value
VU			Interpretation results in "true", if any of the selected VU in any of the selected channels is higher or equals the programmed reference value.
	../PRM/SEL	NONE, IN, OUT, CMP, LIM	Any combination of the values listed is possible. LIM = Limiter CMP = compressor
	../PRM/CHAN	A / B / A,B	Selects channel A, or channel B, or channel A and B
	../PRM/DB	[dB]	VU reference value
CTL_IN			Interpretation results in "true", when the selected control input is activated.
	../PRM/IDX	1, 2	Selects a control input
ERRFLAG			Interpretation results in "true", when one of the selected flags is set. Any combination of the flags listed is possible.
	../PRM/ DEVICE		Mask for device state/error flags
	../PRM/IN_A		Mask for input channel A state/error flags
	../PRM/IN_B		Mask for input channel B state/error flags
	../PRM/OUT_A		Mask for output channel A state/error flags
	../PRM/OUT_B		Mask for output channel B state/error flags
MEMFLA G			Interpretation results in "true", when the actual state of the selected memo flags resembles the reference pattern.
	../PRM/MASK	NONE, 1...16	Selects memo flags to be interpreted (listing)
	../PRM/VALUE	NONE, 1...16	Defines the expected reference pattern for memo flags
PRESET			Interpretation results in "true", when the actual preset is identical to a selected preset.
	../PRM/DIRTY	ON, OFF	Selection is also valid, when parameters have been changed (dirty)
	../PRM/USER	NONE, 1...6	List of selected user presets

	../PRM/ OWNER	NONE, 1, 2	List of selected owner presets
	../PRM/FACT	NONE, 1, 2	List of selected factory presets

Firmware Upgrade

The firmware of RCM-26 remote amps is stored in a FLASH-memory chip. This technology has been chosen to be able to provide the users with new software without the hassle of physically exchanging memory chips inside of a remote amplifier. Using IRIS-Net, upgrading the firmware is possible via the CAN Remote Control Interface. In this way you can install new firmware and future software extensions to always keep your Remote Amplifier System up-to-date.
The RCM-26 firmware is divided into a part for basic amplifier functions (e.g. power on/off, CAN communication) and a part for extended functions (e.g. signal processing). Even if the firmware update procedure does not finish successfully, basic amplifier functions stay functional and the update procedure can be repeated.

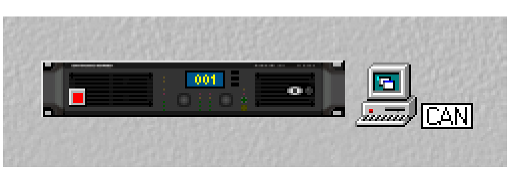
Caution!
Upgrading the firmware is always a very sensible procedure – comparable to updating the OS in the FLASH-memory of a PC. Therefore, obeying the following precautions and instructions is absolutely mandatory:
Consequences

1. Simultaneously upgrading the firmware of more than four remote amplifiers is not recommended.
2. Only connect the remote amps to the CAN Remote Control network that are to be updated. Disconnect any other remote amps from the CAN-bus during the upgrade. Make sure to carefully mind all regulations for the CAN Remote Control network, especially the 120 Ω termination at both ends of the bus.

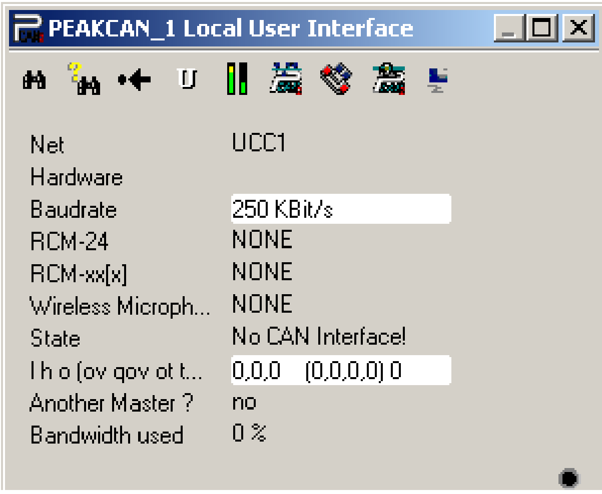
HOW TO UPGRADE THE FIRMWARE

Necessary preparations

1. Connect the desired remote amp(s) via CAN-bus to your PC.
2. Start the IRIS-Net software and open your project. Your remote amps and the icon of a PC with CAN-label should appear on your screen. The PC-icon represents the CAN-interface of your PC or notebook.



3. Double-clicking onto the PC-icon opens the CAN-interface window. CAN-bus status and connected remote amps are displayed. This window display is available in off-line mode.



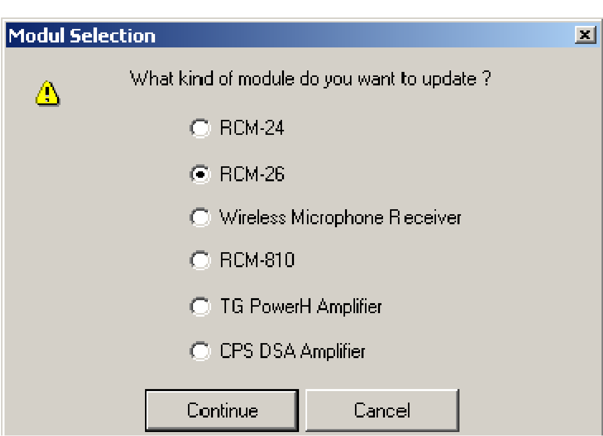
Make sure to check the following parameters before upgrading:

Element	Description
Baud rate	Indicates the set baud rate. Normally you don’t have to change the system’s baud rate for upgrading.
RCM-xx[x]	Indicates the addresses of the remote amps connected. Make sure that the addresses shown are only the ones of remote amps that you want to upgrade.
State	Indicates the CAN-interface status. This has to read “OK”. Otherwise, starting the firmware upgrade is not permissible.
I h o (...	Indicates different error flags. Under no circumstance the first 3 digits may rise. Clicking into the white field and entering “0” resets the error flags.
Bandwidth used	Indicates the used bandwidth of the CAN-bus in percent. Make sure to check that the CAN-bus is not too busy, i.e. high data traffic.

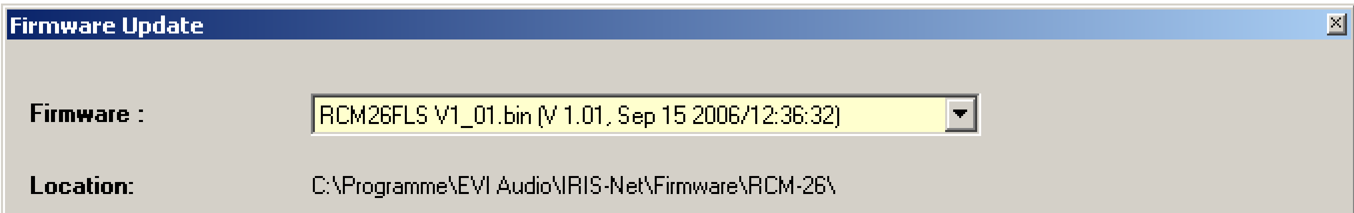
You can update the firmware of the remote control module or the firmware of the amplifier itself.

Firmware update of the Remote Control Module

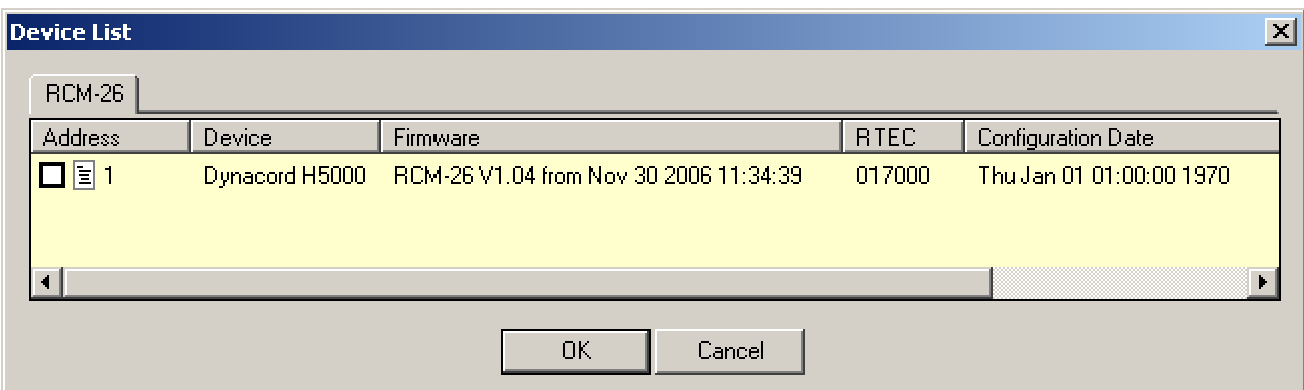
1. The CAN-interface window provides a toolbar (top line). Clicking onto the U-icon (Update) opens a Module Selection dialog.
2. Select RCM-26 and click onto "Continue".



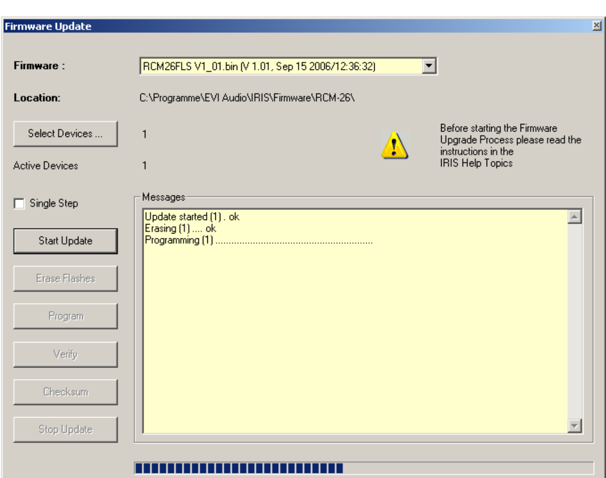
3. The actual firmware file including version number and date is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date remote amplifier firmware version. The corresponding file is located in the directory: \IRIS-Net\Firmware\RCM-26. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.



4. Click onto the button “Select Devices...” to open a list of all remote amps connected. Select the amp(s) that you want to update and click the “OK” button. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus. When performing the firmware upgrade for the first time, connecting only a single amplifier is recommended to become familiar with the upgrading procedure.



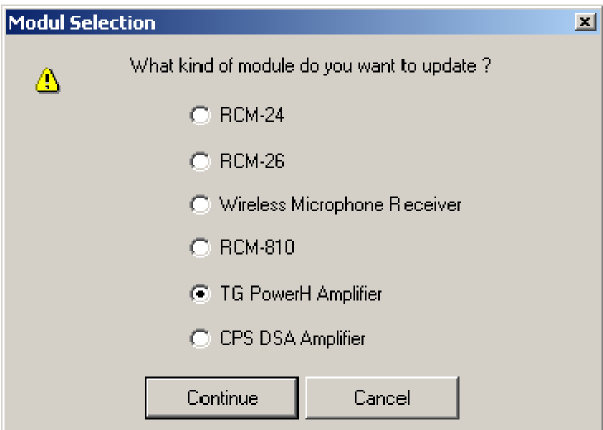
5. The addresses of the selected remote amp(s) are shown in the firmware update window on the right side next to the button “Select Devices...” and in the line “Active Devices”. Clicking onto “Start Update” starts the upgrade procedure. The single steps of the update are shown in the “Messages” window. The progress of some parts of the upgrade which take a little longer is indicated through dots behind the corresponding name. The message “ok” has to appear at the end of each line. The following example shows how to upgrade the firmware of the remote amp with the address 1 to firmware version V 1.01.



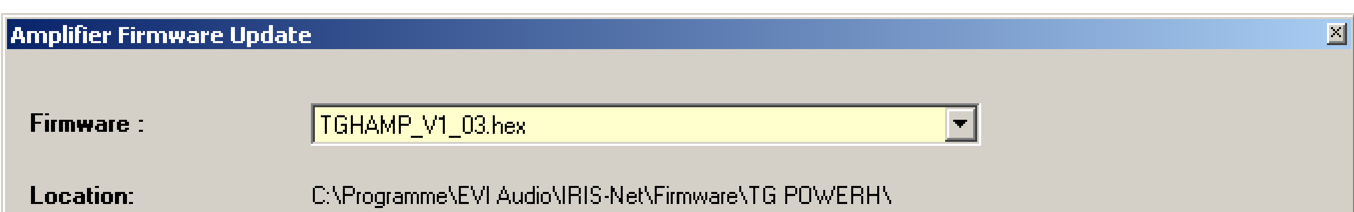
6. The message “Finishing .. ok” indicates that upgrading has been successful. The remote amp(s) are reset. Afterwards they are again ready for operation. The upgrade procedure is finished and you can close the dialog window or proceed with upgrading other remote amps.

Firmware update of the amplifier

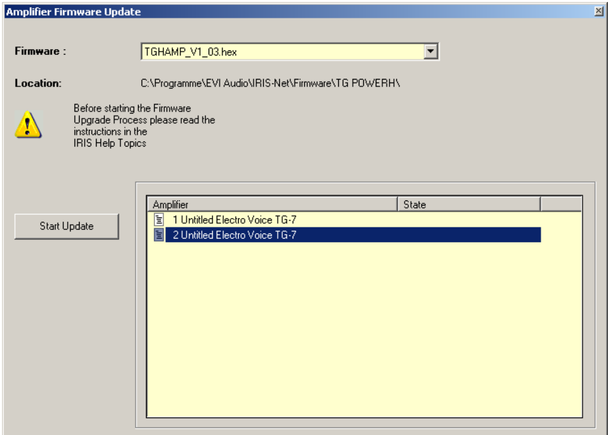
1. The CAN-interface window provides a toolbar (top line). Clicking onto the U-icon (Update) opens a Module Selection dialog.
2. Select TG PowerH Amplifier and click onto Continue



3. The actual firmware file including version number and date is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date remote amplifier firmware version. The corresponding file is located in the directory: \IRIS\Firmware\TG POWERH. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.



4. Select the amplifier that you want to update and click the “OK” button. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus.
5. Clicking onto “Start Update” starts the upgrade procedure. The single steps of the update are shown in the “Messages” window. The progress of some parts of the upgrade which take a little longer is indicated through dots behind the corresponding name. The message “ok” has to appear at the end of each line. The following example shows how to upgrade the firmware of the remote amp with the address 2 to firmware version V 1.03.



6. The message “Finishing .. ok” indicates that upgrading has been successful. The remote amp(s) are reset. Afterwards they are again ready for operation. The upgrade procedure is finished and you can close the dialog window or proceed with upgrading other remote amps.

– The line “Active Devices” indicates which of the selected remote amps are still to be updated. Amps for which the update process timed out are taken off the list. These devices are still capable of receiving upgrade commands. However, the software does not wait for acknowledgements of the concerned amps any longer.
– If the IRIS-Net-software recognizes an error or “Time Out” during upgrading, it automatically switches to “Single Step” mode, which offers the possibility to repeat the upgrade in single steps. If a “Time Out” message is displayed while upgrading is in progress, under no circumstance switch off any amps!
– As soon as “Single Step” is checked off, all buttons below the single step field become active. The upgrade can now be performed manually, step- by-step in the sequence as described below. If one of the commands does not finish “ok”, you have to restart the upgrade procedure from the beginning.

Step	Description
Start Update	Activates update mode for the selected devices. The messages window shows “Update started (addresses)” .. and after a short period of time “ok”.
Verify	Compares the firmware installed in the remote amps with the selected firmware file. The messages window shows “Verifying (addresses)”...a progression-bar indicates the approximate duration of the process. Detected differences are indicated at the end of the process, e.g. “done, Errors detected for...”. If no errors time-outs are detected, you can proceed with the update.
Erase Flashes	Deletes the actual firmware and clears the FLASH-memory of a remote amplifier. The messages window shows “Erasing (addresses)”..... and after a short period of time “ok”.
Program	Loads the new firmware into the FLASH-memory of a remote amplifier. The messages window shows “Programming (addresses)"........ A progression-bar indicates the approximate duration of the programming. “ok” appears in the message window after some time.
Checksum	Evaluates the checksum of the newly installed firmware. The messages window shows “Checksum (addresses)” .. and after a short period of time “ok”. This is a short form of the “Verify” process.
Stop Update	Finishes upgrading. The messages window shows “Finishing (addresses)" .. and after a short period of time “ok”. The remote amps quit the update mode and start in nor- mal mode. Now, you can exit the upgrade dialog or proceed with upgrading other remote amps.

- If “Time Out” errors still occur during the programming, repeat the procedure in single step mode in the following sequence: Start Update - Program.
- If the checksum evaluation shows errors, repeat the entire upgrade procedure. Don’t forget to uncheck “Single Step” mode, for the upgrade to run automatically.

RCM-28

The RCM-28 Remote Control Module is a two-channel digital controller module for live sound reinforcement, PA and fixed installation applications. The module can be used in Electro-Voice Tour Grade and DYNACORD PowerH Amplifier models. Installing the RCM-28 turns a conventional amplifier into a remote amplifier, which, at any time, provides a complete overview of the overall system status and control of all system parameters.

RCM-28 modules allow the integration of amplifiers into a OMNEO network with up to 100 devices in a single subnet without additional hardware. This offers the possibility to control and monitor an entire sound system from one or more PCs using the IRIS-Net - Intelligent Remote & Integrated Supervision - software package. All operational states, such as power-on status, temperature, modulation, limiting, activation of protections, deviation from the load impedance, etc., are centrally registered and displayed in IRIS-Net. This provides the possibility to react, and to selectively intervene even before critical operational states arise. Programming an automatic reaction, when specific thresholds are being exceeded or fallen below, is also possible.

With an RCM-28 installed, the integrated impedance testing function allows very precise monitoring of the connected loudspeaker systems. The impedance testing function utilizes the internal test tone signal generator and voltage/current measuring to determine the impedance of the loudspeaker systems including crossovers and cables over the entire frequency range. IRIS-Net plots an impedance curve of the measured impedance that can be compared with a previously stored reference curve at any time. This reveals even smallest loudspeaker defects or deficiencies instantly.

Parameters, such as power on/off, level, muting, filters, etc. can be controlled in real-time and stored in the amplifier.

Besides controlling and monitoring amplifiers, the RCM-28 also offers all conventional signal processing functions, such as parametric equalizers, frequency crossovers, delays, Peak Anticipation and TEMP limiters. Beyond that, linear-phase FIR-filters and digital loudspeaker protection algorithms are available to optimize the amplifiers and loudspeaker system. All DSP-settings can be freely edited and stored in user presets directly on the module. In the event of network failure or loss of power, all settings (filters, delay, level, etc.) stay intact, independent of the control by the network.

Additionally, the RCM-28 provides a control port with freely programmable control inputs and control outputs. Control inputs (GPIs) allow the connection of switches. IRIS-Net offers the possibility to program a variety of logic functions for the inputs (e.g. switching to an alarm-preset with maximum energy in the speech area). Control outputs (GPOs) allow the connection of external components, which, for example, are used to signal specific states to peripheral equipment. Consequently, an amplifier with a RCM-28 module installed corresponds to highest safety requirements.

The RCM-28 has been designed with uncompromising audio quality in mind. Analog audio inputs (internally, pre or post fader), an AES3 (AES/EBU) digital audio input with XLR-type connector and an OMNEO port for connection to OMNEO or Dante audio networks are provided.

OMNEO

OMNEO is a media networking architecture for professional applications. Using standard IP Ethernets, media products that integrate OMNEO can be assembled into networks of 2 to 10,000 cooperating devices that exchange studio-quality synchronized multichannel audio and share common control systems.
OMNEO's media transport technology is Audinate's Dante, a high-performance standards-based, routable IP media transport system. OMNEO's system control technology is OCA, for Open Control Architecture, an open public standard for control and monitoring of professional media networks.
For additional information, please see the OMNEO Resource Guide available at www.electrovoice.com or www.dynacord.com

Remote Amplifiers

The Remote Control Module RCM-28 can be used in following power amplifiers:

DYNACORD POWERH SERIES

– H 2500 2 x 1450 W / 4 Ohm or 2 x 2000 W / 2 Ohm
– H 5000 2 x 2500 W / 4 Ohm or 2 x 3500 W / 2 Ohm

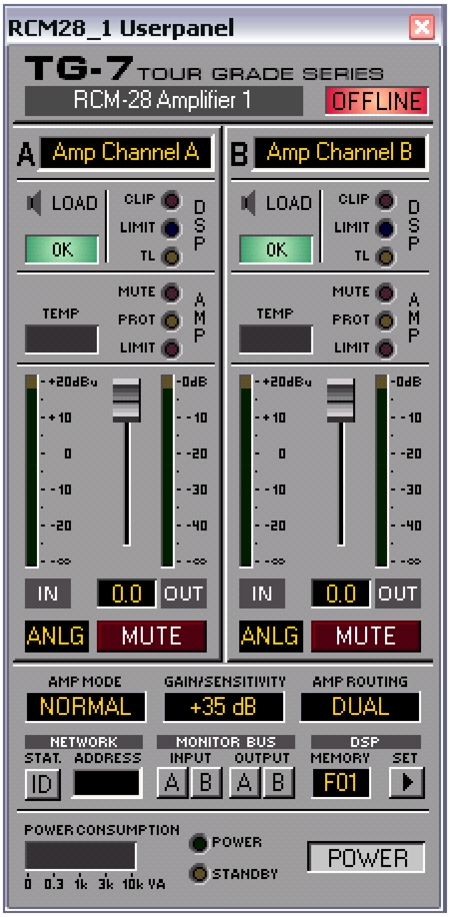
ELECTRO-VOICE TOUR GRADE SERIES

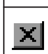
– TG-5 2 x 1450 W / 4 Ohm or 2 x 2000 W / 2 Ohm
– TG-7 2 x 2500 W / 4 Ohm or 2 x 3500 W / 2 Ohm

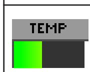
The power amplifiers mark a milestone in the design and the production of high-performance power amplifiers. The innovative 3-stage Grounded Bridge Class H Topology with ”floating” switching power supply unit offers very high and stable output with extreme high efficiency on an extremely high performance level at minimum weight.
PowerH / Tour Grade Series amplifiers are ideal for driving professional touring, high-end Concert-Sound and Pro-Sound applications. Next to classical protections, this new design employs the multi-stage ATP system (Advanced Thermal Protection) for the first time, which in most cases prevents the power amplifier from switching off when the temperature exceeds a critical level. The newly designed MCS system (Mains Current Supervision) prevents power amplifier breakdown caused by the activation of the automatic circuit breaker. For this, among other things, the MCS system uses the highly precise measurement of the RMS value of the actual mains current consumption. Information about the status of the power amplifier and its internal protections is provided on a LC-display. By utilizing the optionally available remote control module that is compatible with IRIS-Net, this power amplifier additionally offers comprehensive remote monitoring and remote control functions plus a universal 2-channel digital audio controller (DSP) including highly precise FIR-filtering.

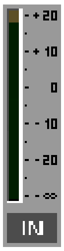
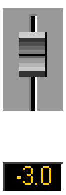
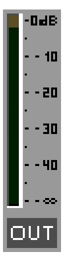
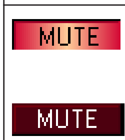
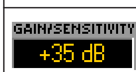
Amplifier Control Panel

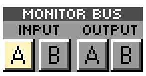
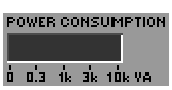
Double clicking with the left mouse button on an amplifier gets you to the Amplifier Control Panel, which provides access to the most important controls and indications of the selected amplifier. Simultaneously opening several Amplifier Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.



Element	Description
TG-7	Amplifier Type (generated during amplifier selection or read from the amp while being on-line)
	Using the left mouse button, click on the Close button to close the Amplifier Control Panel.
	A name can be assigned to each amplifier to specify its use or position. Click on the gray-shaded entry field below the Amplifier Type field and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: The amplifier name is the network-wide OMNEO device name and will be used for identifying a device in the network. Make sure that the name occurs only once in a network. Each device must have a different name.** **HINT: Entering amplifier names is also possible within the Setup & Control Panel on the Config & Info page.** **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
ONLINE OFFLINE	The Online / Offline indicator signals whether the selected amplifier is currently communicating with IRIS-Net. The red OFFLINE indicator signals that the corresponding amplifier is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding amplifier is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.

	The amplifier channels are named channel A and B. A name can be assigned to each channel to easily identify its allocation and use. Using the left mouse button, click in the entry field and enter the desired name for the channel. Press Return on the keyboard to acknowledge your entry. **HINT: Entering channel names is also possible within the Setup & Control Panel on the Config & Info page.**
	The CLIP indicator lights whenever the signal of the internal signal processor clips. The signal processor's headroom is 12 dB, which is no problem when using normal filter settings. However, when drastically increasing the level of several adjacent or overlapping filters, distortion of high-level signals may occur, which the CLIP indicator indicates. In that case reducing the signal-level or trying a bit more moderate equalizer setting is recommended.
	The LIMIT indicator lights whenever the digital limiter of the corresponding channel is activated, e.g. when the signal level exceeds the specified threshold and the output level is being limited to this value.
	The TL indicator lights when the loudspeaker TEMP limiter of the corresponding channel is activated.
	The LOAD indicator shows whether the load connected to the amplifier output is within the allowable range or if short-circuit or line interruption has occurred. The green OK-indication signals that the connected load is between the specified lower and upper limit values. These values are set in the Setup & Control Panel in the Load screen. The red OPEN indication signals line interruption. It lights whenever the connected load exceeds the upper limit value. The red SHORTED indication signals short-circuit at the amplifier output. It lights whenever the connected load falls below the lower limit value. **HINT: The connected load is monitored continually as soon as a signal with a voltage of > 150 mV is present at the output. Calculation of signal levels below that threshold is not possible and the indicator shows the last acquired state.**
	The TEMP display shows the amplifier's internal temperature as a graph. The indicator lights green whenever the amplifier is operated in its nor- mal operational temperature range. The indicator lights yellow whenever the amplifier builds up heat because of continuous high output. However, since the internal fans provide sufficient ventilation there is no risk of thermal overload in this state. As soon as temperature indication changes to red, reducing the output level is strongly recommended. Otherwise the amplifier might cease operation because of thermal overload.
	The MUTE indicator lights when the amplifier is muted. This occurs e.g. during speaker switch-on delay or when the amplifier is switched off by the protection circuit (see below).
	When the red PROT indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated However, a lit PROT LED does not necessarily mean that the signal path gets switched off. The differentiated protections concept of the power amplifier results in several protection circuits being activated one after another, which ensures that under normal circumstances the power amplifier will stay in the safe and stabile operating range. In case the amplifier needs to be switched off to prevent power amplifier and connected speaker systems from being damaged, this is indicated by the PROT and MUTE LEDs being lit simultaneously.

	The LIMIT indicator lights as soon as the internal dynamic limiter is activated, which is the case when the amplifier is operated at maximum out- put. Short-term blinking is not a problem, since the internal limiter controls input levels of up to +20dBu down to a distortion rate of approximately 1%. However, if this indicator lights permanently, reducing the output level is strongly recommended to protect the connected loudspeaker systems from being damaged by capacity overload.
	The Input Level Meters provide indication of the corresponding audio levels at the amplifier inputs in dBu. The amplifier’s nominal input level is +6dBu, the maximum level can be as high as +21dBu. In general, it is recommended that the amplifier be operated in a range between 0 and +10dBu. Only signal peaks should be at higher levels.
	The level controls are for adjusting the overall amplification of the corresponding amplifier channel. Setting the level controls to a value between 0dB and -6dB provides full output capacity. The numerical field below the level controls indicates the set level, by which the output amplification is attenuated, in dB.
	The Output Level Meters provide indication of the corresponding audio levels at the amplifier outputs. Indication in dB is relative to amplifier full-modulation. A 0dB output level (full-modulation) is indicated in yellow.
	The currently used audio input (ANLG, AES or OMN..) is indicated.
	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	AMP MODE indicates the operation mode of the power amplifier blocks. Possible settings are NORMAL and BRIDGED. Switching the amp mode is only possible locally at the power amplifier, details can be found in the amplifiers owner's manual.
	GAIN/SENSITIVITY displays the amplifiers gain or input sensitivity. **IMPORTANT NOTE: A RCM-28 firmware update with version V2.0.10 or higher will enable the Sensitivity switch on the amplifier rear panel. Please make sure that the Sensitivity switches of all amplifiers in your system are set correctly. In previous firmware versions the amplifier gain was fixed to +35 dB if a RCM-28 module was installed.**
	AMP ROUTING shows how the audio inputs handle the input signals. Possible settings are DUAL and PARALLEL. Switching the amp routing is only possible locally at the power amplifiers, details can be found in the amplifiers owner's manual.

	Clicking this switch activates the LOCK indicator on the amplifier's rear panel and all amplifier front panel LEDs, as well as in the amplifier's front panel window in the IRIS-Net software. This function is meant for identifying or searching an amplifier in a large system setup.
	The number field indicates the set amplifier object number. This is an ascending number which will be assigned automatically when the amplifier will be created in the IRIS-Net work sheet. It is the same number as shown on the device front panel in IRIS-Net.
	These buttons allow assigning amplifier channels to the RCM-28 OMNEO output 3, which is reserved for monitoring. This allows monitoring any amplifier input or output signals within an installation. INPUT A / B selects the corresponding input signal while OUTPUT A / B allows switching between the output signals of channels A and B. Simply click on an amp channel's icon to select it for monitoring. The corresponding channel is assigned to OMNEO output channel 3 of the amplifier. Any previous selection is simultaneously canceled, so that only the actually selected amp channel can be monitored. Clicking the button of an active amp channel separates the channel from the monitor bus. For monitoring you have to select the desired amplifier output 3 in the Network View dialog and assign it to an OMNEO/Dante receiver, e.g. the Dante Virtual Sound Card.
	This field indicates the active factory or user preset. Each remote amp has two factory setting F01 (linear thru) and F02 (DSP Bypass) offering linear settings and thirty user-programmable presets U01...U30 for storing random user data. Loading and saving presets is done in the Setup & Control window.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all amplifier- and DSP-parameters, control and monitoring functions plus additional function groups.
	POWER CONSUMPTION indicates the current power consumption of the power amplifier in VA.
	This soft-key allows switching an amplifier on or off. The STANDBY and POWER indicators signal the actual operational status. The Config & Info window allows programming individual power-on delays for each amplifier. **HINT: The power-on delay defaults to „Object No. * 150 ms“. For object no. 8 the power-on delay default would be for example: 8 *150 ms = 1200 ms.**
	These indicators show the amp's actual operational status. STANDBY lights whenever the amplifier is in stand-by mode. POWER lights whenever the amplifier is powered-on and ready for operation. If neither one of the indicators lights, the amplifier is either off-line or powered-off.

Setup & Control

The Setup & Control window allows configuring all amplifier parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

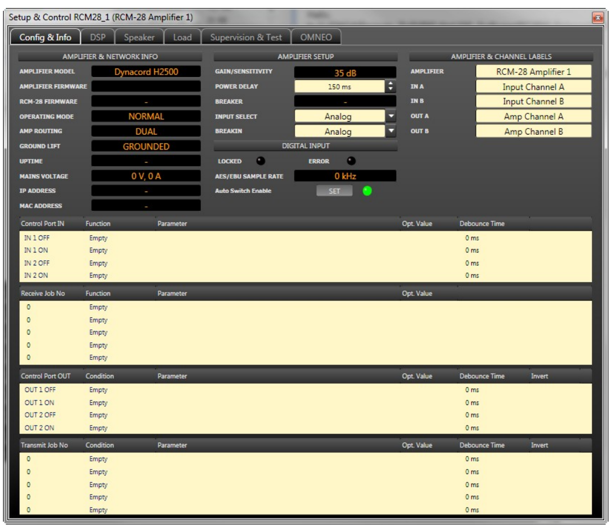
Window	Description
Config & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Filter, Delay, X-Over, Limiters) of the amplifier.
Speaker	This page allows loading and displaying speaker data.

Load	This page provides access to several settings for impedance/load monitoring and impedance testing.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.
OMNEO	This page provides access to the OMNEO interface and routing parameters.

Clicking on the „SET“ soft key in the Amplifier Control Panel opens the Setup & Control window.

Config & Info

The Config & Info window provides information and basic settings for the selected amplifier. Additionally, editing labels is possible in this window.
To select the page click on the Configuration & Information tab in the Setup & Control Window.



Amplifier Info

Element	Description
AMPLIFIER MODEL	Shows the amplifier type
AMPLIFIER FIRMWARE	Shows the amplifier‘s software version number (operating system, firmware)
RCM-28 FIRMWARE	Shows the remote control module‘s software version number (operating system, firmware)
OPERATING MODE	Shows the amplifier’s operating mode. The power amplifier can be operated in NORMAL or BRIDGED mode.
AMP ROUTING	Shows how the amplifier's audio inputs handle the input signals. Possible amp routings are DUAL and PARALLEL.
GROUND LIFT	Shows the setting of the amplifier's GROUND LIFT switch. Possible settings are GROUNDED and UNGROUNDED.

UPTIME	Shows the uptime of the amplifier (standby not included) since the last reset of the Event Log. For further details about the amplifiers's Event Log please refer to the owner's manual.
MAINS VOLTAGE	Shows the mains voltage and the mains current consumption.
IP ADDRESS	Shows the IP address of the remote module‘s primary interface. The IP address of the secondary interface can be seen in the OMNEO page.
MAC ADDRESS	Shows the MAC address of the remote control modul's primary interface. The MAC address of the secondary interface is one number higher and can be seen in the OMNEO page.

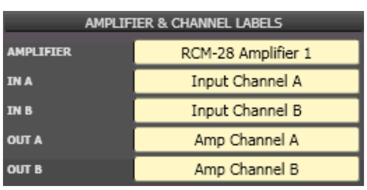
Amplifier Setup

Element	Default	Range	Description
GAIN/ SENSITIVITY			GAIN/SENSITIVITY displays the amplifiers gain or input sensitivity. **IMPORTANT NOTE: A RCM-28 firmware update with version V2.0.10 or higher will enable the Sensitivity switch on the amplifier rear panel. Please make sure that the Sensitivity switches of all amplifiers in your system are set correctly. In previous firmware versions the amplifier gain was fixed to +35 dB if a RCM-28 module was installed.**
POWER DELAY	Object No. * 150 ms	50...6350 ms 50 ms Steps	Allows programming an amplifier’s power-on delay. Setting different delay times is recommended to prevent the mains fuse from blowing when powering on several amps at the same time.
BREAKER			Show the current setting of the amplifier's Mains Circuit Breaker Protection. For further details about this protection please refer to the owner's manual.
INPUT SELECT	Analog	Analog, AES/ EBU, OMNEO	Allow selecting the audio input to be amplified. The analog audio input (ANALOG), the digital audio input (AES/EBU) or the OMNEO audio input are available. **HINT: The value of the property “AudioInput” corresponds to the currently used audioinput. You can write the value “ANLG” to this property to select the analog audioinput. You can write the value “AES” to this property to select the digital audioinput. You can write the value “NET” to this property to select the OMNEO networkinput.**
BREAKIN	Analog	Analog, AES/ EBU, Auto	Allows selecting the audio input of the amplifier to be output via the two OMNEO audio network Tx channels. The analog audio input (Analog) or the digital audio input (AES/EBU) are available. Select Auto to output the digital audio signal and switch from digital input to analog input automatically if the digital input signal is not OK.

Digital Input

Element	Default	Range	Description
			Indicates if the digital audio input is synchronized to the input signal (LOCKED LED on) or if synchronization was not successful (ERROR-LED on). **HINT: The current state of the digital audio input is available via the properties "State Flags. AES Locked" and "State Flags. AES ERROR" also.**
AES/EBU SAMPLE RATE		32...192 kHz	Shows the sample rate of the AES/EBU input signal.
Auto Switch Enable			Press the SET button if the audio input should switch from digital input to analog input automatically if the digital input signal is not OK. **The value of the property "Audio Input Switching. AES Fail. Enable" corresponds to this control.**

Amplifier & Channel Labels

Element	Description
	The labels of an amplifier and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately reflected in the different panels and windows (amplifier control panel, flow diagram, and overview). **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

SWITCHING AUDIO INPUTS AUTOMATICALLY

RCM-28 remote amplifiers allow automatic switching from the AES feed to the analog input in the case of failure of the AES signal or input. This feature allows a redundant analog signal to be fed to the amplifier and automatically switch over to the „backup“ signal without user intervention.

Open the Modify Properties dialog from the context menu of the RCM-28 remote amplifier in the IRIS-Net worksheet. Following table lists the relevant properties for automatic audio input switching.

Property	Default	Range	Description
Audio Input Switching. AES Fail. Enable	0	0, 1	Set the value to 1 to enable the auto fall back function which switches from AES to analog input in case of an AES error. This property corresponds to the „Auto Switch Enable“ control found in the Config & Info window.
Audio Input Switching. AES Fail. Time	1 s	0 to 120 s	Configures the duration the AES error had to be present on the input to trigger the AES / analog switching.
Audio Input Switching. AES Good. Enable	0	0, 1	Setting this property to 1 enables the automatic switching from analog input to AES input in case the AES signal is OK. **HINT: This works only if the values of both properties – Audio Input Switching. AES Fail. Enable and Audio Input Switching. AES Good. Enable – are set to 1.**

Audio Input Switching. AES Good. Time	5 s	0 to 120 s	Configures the duration the AES signal had to be in „Locked“ state, to switch back from analog to the AES input.
Audio Input Switching. AES Ok Flag	0	0, 1	Shows that the AES signal is locked and without errors for longer than the configured Audio Input Switching. AES Good. Time.
Audio Input Switching. AES Selected Flag	0	0, 1	Shows that the AES input is used as amplifier audio input.
Audio Input Switching. Counter	0	-	Counts how often the Auto Switch ganged the input.

Control Port

A control port offering two control inputs and two control outputs is located on the amplifier’s rear panel. The functions of these inputs and outputs can be programmed in a variety of ways. For example, the control inputs (GPI) can be used for power-on / stand-by or preset switching as well as for changing parameters. The control outputs (GPO) are for signaling internal statuses. They can directly trigger LEDs, control lamps or relays. In the Supervision & Test window the states of the control inputs are displayed and you have the possibility to switch the control outputs manually. For more information and electrical specifications of the control port, please refer to the amplifier‘s manual.

Control Inputs: Each status change of a control input can trigger a function. Different functions can be assigned for the opening (OFF) or closing (ON) of a contact.

Example:

Control Port IN	Function	Parameter	Opt. Value	Debounce Time
IN 1 OFF	Power	on		0 ms
IN 1 ON	Power	off		0 ms
IN 2 OFF	Preset	U02		0 ms
IN 2 ON	Preset	U03		0 ms

This example shows the programming of two control inputs where IN1 switches the amplifier on or off and IN2 selects presets U02 or U03.

- IN1 OFF: Power off (opening the contact of control input 1 switches the amplifier to stand-by)
- IN1 ON: Power on (closing the contact of control input 1 switches the amplifier on)
- IN2 OFF: Preset U02 (opening the contact of control input 2 selects preset U02)
- IN2 ON: Preset U03 (closing the contact of control input 2 selects preset U03)

Element	**Default**	**Range**	**Description**
Control Port IN		IN 1 OFF IN 1 ON IN 2 OFF IN 2 ON	This provides a listing of the two control inputs and their statuses ON and OFF. The entries in the corresponding lines specify the action when closing (ON) or opening (OFF) a contact.
Function	(empty)		This column allows assigning functions to a control input’s statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.

Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. Following a status change the assigned function is initiated after the set time interval has passed.

Control Outputs: Internal status changes inside of the amplifier, such as operational faults, alerts when exceeding parameter limits, and internal operational statuses can be signaled to external systems or central control units.
Example:

Control Port OUT	Condition	Parameter	Opt. Value	Debounce Time	Invert
OUT 1 OFF	Power			0 ms	X
OUT 1 ON	Power			0 ms	
OUT 2 OFF	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUTA.Z_MIN,OUTA.Z_MAX,OUTA.THRM_HDR		0 ms	X
OUT 2 ON	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUTA.Z_MIN,OUTA.Z_MAX,OUTA.THRM_HDR		0 ms	

This example shows the programming of two control outputs where OUT1 signals whether the amplifier’s power is switched ON or OFF while OUT2 signals faulty operation.

– OUT1 OFF: Invert Power (control output 1 is open when the amplifier’s power is switched off / stand-by mode)
– OUT1 ON: Power (control output 1 is closed when the amplifier’s power is switched on)
– OUT2 OFF: Invert Error flag (control output 2 is open when no faults have occurred)
– OUT2 ON: Error flag (control output 2 is closed when operational faults according to the parameter list have occurred)

Element	**Default**	**Range**	**Description**
Control Port OUT		OUT 1 OFF OUT 1 ON OUT 2 OFF OUT 2 ON	This provides a listing of the two control outputs and their statuses ON and OFF. The entries in the corresponding lines specify which status results in the closing (ON) or opening (OFF) of a contact.
Condition	(empty)		This column allows assigning internal events (conditions) to a control output’s statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. An event is signaled following an internal status change and after the specified time interval has past.
Invert	(empty)	(empty) / X	This column allows entering whether a status is signaled when the specified Condition is „true“ (no entry) or „false“ (click „X“ to signal an inverted state).

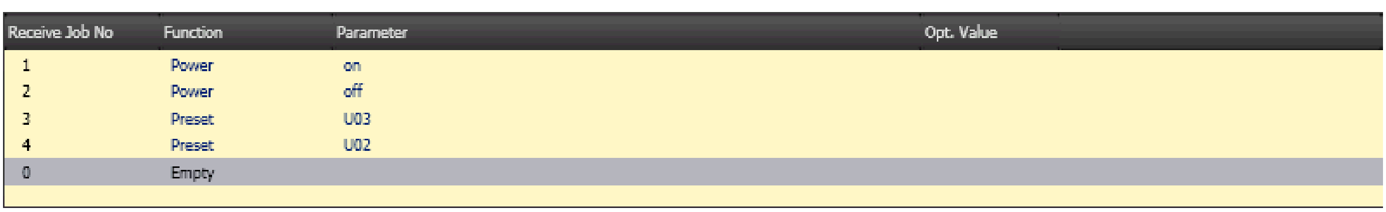
Jobs

For amplifiers to be able to communicate with each other, it is possible to send and receive Job Codes. In principle, a job code is a function number that an amplifier transmits via OMNEO and that is received and interpreted by another or several other amplifiers. Each amplifier is capable of transmitting and receiving up to 5 different job codes. Programming job codes is nearly identical to the programming of control inputs and outputs.

Caution!

Job codes make the remote triggering of functions from one amplifier to another very convenient by sending remote commands over the network. This can reduce cabling and make installation easier. The job code transmission mechanism in the RCM-28 uses UDP, which does not have the same delivery confirmation and retransmission characteristics as TCP. Due to this, while the operation and transmission of these codes is generally very stable and reliable, there is a very small chance that a message may be lost due to heavy network traffic. If job code or GPI functionality is needed for a mission-critical or life safety application, it is recommended to use a network with enough free bandwidth and make sure that no other heavy network traffic can occur, or to use direct hardwiring to GPIs on each amplifier instead of job codes.
Consequences

Receive Jobs: A receive job is a function that is carried out as soon as the corresponding function number (the Receive Job Code) is received.
Example:



Receive Job No	Function	Parameter	Opt. Value
1	Power	on	
2	Power	off	
3	Preset	U03	
4	Preset	U02	
0	Empty		

This example shows the programming of four Receive Jobs. Jobs No. 1 and 2 switch the amplifier's power on or off while jobs No. 3 and 4 select presets U03 or U02. The fifth Receive Job has not been configured.

– Receive Job Nr. 1: Power on (receiving Job Code 1 switches the amplifier's power on)
– Receive Job Nr. 2: Power off (receiving Job Code 2 switches the amplifier into stand-by mode)
– Receive Job Nr. 3: Preset U03 (receiving Job Code 3 selects preset U03)
– Receive Job Nr. 4: Preset U02 (receiving Job Code 4 selects preset U02)

Element	Default	Range	Description
Receive Job No	0	1...1023	Here, you can specify which incoming job code numbers a specific amplifier recognizes. Entering random numbers between 0 and 1023 is possible.
Function	(empty)		This column allows assigning an individual function to each job code received. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.

HINT: Programming identical control functions or receive jobs for several amps is easily accomplished by creating a group that includes all the desired amps and afterwards perform the programming in the group's Configuration& Information dialog. All settings are automatically applied to all amplifiers of that group, which saves time and effort and additionally reduces the risk of programming errors.

Transmit Jobs: Transmit Job defines a function number that is sent as soon as a specific internal event (condition) occurs in the amplifier.

Example:

Transmit Job No	Condition	Parameter	Opt. Value	Debounce Time	Invert
1	GPI	IN1		0 ms	
2	GPI	IN1		0 ms	
3	GPI	IN1		0 ms	
4	GPI	IN1		0 ms	
0	Empty			0 ms	

This example shows the programming of four Transmit Jobs. Jobs No. 1 and 2 are triggered by control input 1. Jobs No. 3 and 4 are triggered by the status signaled from control input 2. The fifth transmit job has not been configured.

– Transmit Job Nr. 1: GPI IN1 (Job Code 1 is transmitted when control input 1 is closing)
– Transmit Job Nr. 2: Invert GPI IN1 (Job Code 2 is transmitted when control input 1 opens)
– Transmit Job Nr. 3: GPI IN2 (Job Code 3 is transmitted when control input 2 is closing)
– Transmit Job Nr. 4: Invert GPI IN2 (Job Code 4 is transmitted when control input 2 opens)

Element	Default	Range	Description
Transmit Job No	0	1...65536	Here, you can specify which job code numbers an amplifier transmits on the occurrence of specific events. Entering random numbers between 0 and 65536 is possible.
Condition	(empty)		This column allows specifying an event (condition) that triggers the transmission of the corresponding job code. Clicking the desired line in the Condition menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here, you can program delay or debouncing times. A transmit job code is sent following a specific event and after the specified time interval has past.
Invert			This column allows entering whether a job code is transmitted when the specified Condition is "true" (no entry) or "false" (click "X" to signal an inverted state).

Input and Receive Job Functions: The following table lists all functions together with their individual settings, which can be triggered via control input or Receive Job.

Function	Parameter	Opt. Value	Function executed
Empty	-	-	None

Power	off on flip		Power Off (Standby) Power On Power-status change (ON to Stand-by and reverse)
Absolute	All DSP parameters	Corresponding Parameter Value (parameter-dependent)	Set the specified absolute parameter value for the selected parameter
Relative	All DSP parameters	Parameter Value Off- set (parameter-dependent)	Changes the actual value of the selected parameter by the specified offset value
Toggle	Parameters with two statuses		Changes the status of the selected parameter (e.g. bypass On / Off)
Preset	U01-U30, F01, F02		Changes a preset to the specified preset number
Memo flag	Set, Clear, Toggle Memo flags 1 - 16		Sets, erases or changes selected memory flags. Up to 16 memory flags are available and simultaneously accessible.
Measurem ent	Generator frequency, Time, Level A / B		Starts measurement with a tone signal of the specified frequency at the levels specified for channels A / B for the selected duration (0 ms = infinite)
Test generator	Channel, Signal type, Frequency, Solo/Pre, Level		Starts the test generator with selected signal type or of the specified frequency at the levels specified for channels A / B for the selected duration (0 ms = infinite)

Output and Transmit Job Conditions: The following table lists all amplifier statuses that can be used for triggering control outputs or for sending Transmit Job Codes.

Function	Parameter	Opt.Value	Invert	Triggering Event/Status Change
Empty	-	-		Not configured
Power			 X	Power On Power Off (Standby)
Absolute	all DSP parameters	Corresponding Parameter Value (parameter-dependent)	X	Set parameter value reached or exceeded Set parameter value declined
Temp	Temperature in °C		X	Set temperature reached or exceeded Set temperature declined

VU	Channel A or B: Input, Output, PEAK Limiter, TEMP Limiter	Level in dB	X	Set level reached or exceeded Set level declined
GPI	IN 1, IN 2		X	Control input 1 / 2 closed (ON) Control input 1 / 2 open (OFF)
State Flag	All internal fault conditions		X	Single or several error flags set None of the selected error flags set
Memo flag	Enable for selected flags as well as bit- pattern of flags 1 - 16		X	Memory flags match the selected bit-pattern Memory flags do not match the selected bit-pattern
Preset	U01-U30, F01, F02		X	Specified preset selected Other than the specified preset selected

DSP
The DSP pages provide overview and access to all DSP parameters of an amplifier. Within this window you can use the Flow Diagram Selector to link to different function groups.

FLOW DIAGRAM SELECTOR
The Flow Diagram Selector can be accessed from any DSP page offering navigation icons within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the currently selected block is highlighted.

A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

Page	Description
IN / RTG / OUT	The signal flow display provides an overview of an amplifier’s DSP settings. This area also includes all controls for the preset location and preset file management.
INPUT PEQ	The Input Parametric EQ page provides access to the two 10-band parametric equalizers of the amplifier inputs.
INPUT DLY	This page allows the programming of delay lines for the amplifier channels A and B.
ARRAY PEQ	The Array Parametric EQ page offers access to the 5-band parametric equalizers of the amplifier outputs.
ARRAY DLY	This page allows the programming of delay lines for the output channels.
SPEAKER PEQ	The Output Parametric EQ page offers access to the two 6-band parametric equalizers of the amplifier outputs for speaker equalization.
SPEAKER XOV	Frequency crossover-filters as well as the parameters trim, polarity and delay for both channels are located in the Output X-Over area.
SPEAKER FIR	This page provides a FIR-Filter for each amplifier channel.

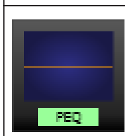
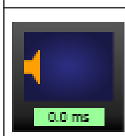
SPEAKERDLY	This page allows the programming of delay lines for the output channels.
SPEAKER LIM	This page provides access to Peak Anticipation limiter and TEMP thermal limiter of each amplifier channel.

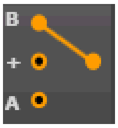
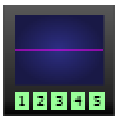
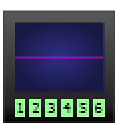
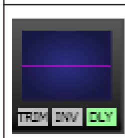
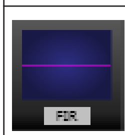
The DSP functions of a remote amplifier can be accessed by clicking onto the SET key in the Amplifier Control Panel followed by a click on the DSP tab in the Setup & Control Window.

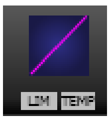
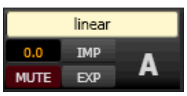
FLOW DIAGRAM

The FLOW DIAGRAM window shows a signal flow diagram, which offers a quick overview of all DSP settings of an amplifier. Routing channels can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for the saving and loading of loudspeaker presets are also accessible from this window.

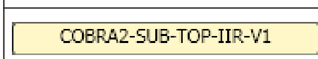
Function blocks

Element	Description
	INPUT PEQ Block: The INPUT PEQ block displays the 10 EQs of the corresponding input channel. The graph shows the frequency response of the EQ block. A single click with the left mouse button onto this block opens the Input Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	INPUT DELAY Block: This displays the Delay of the input channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of amount of delay being applied. A single click with the left mouse button onto this block opens the Input Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other Input Delay block within the same project.

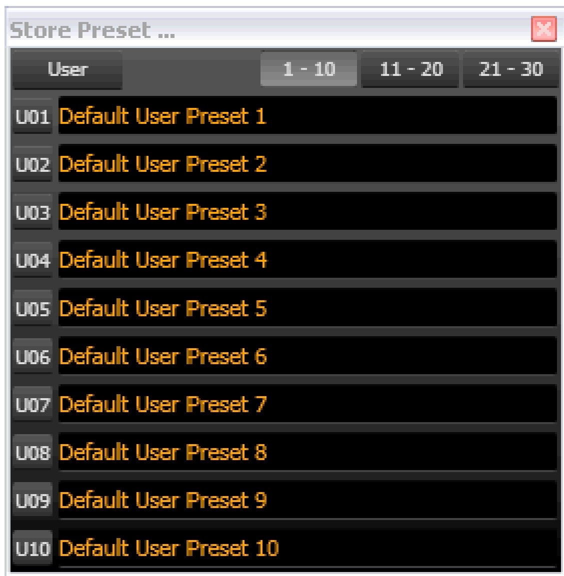
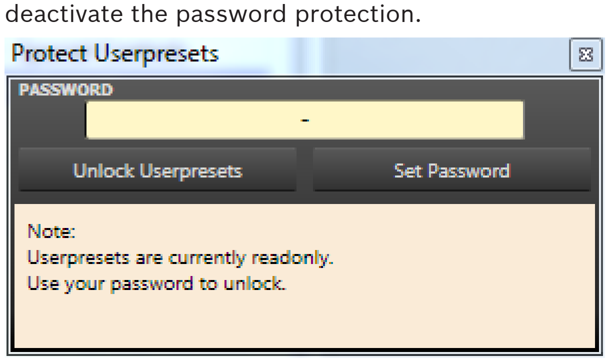
	ROUTING Block: Here you can assign the output channel routing. The circles next to A and B allow selection of the input signal for the corresponding output channel. The circle next to the + allows selection of the summed input signal for the corresponding output channel. A click with the right mouse button onto routing block opens the Copy & Paste menu of all DSP settings, which allows copying all DSP parameters of an amplifier to any other RCM-28 amplifier within the same project.
	ARRAY PEQ Block: The ARRAY PEQ block displays the 5 Array EQs of the corresponding output channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Array Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding ARRAY PEQ block to any other EQ block within the same project.
	ARRAY DELAY Block: This displays the Array Delay of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate amount of delay being applied. A single click with the left mouse button onto this block opens the Array Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other ARRAY DELAY block within the same project.
	SPEAKER PROCESSING PEQ Block: The SPEAKER PROCESSING PEQ block displays the 6 Channel EQs of the corresponding output channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Out-put Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Speaker EQ block to any other EQ block within the same project.
	SPEAKER PROCESSING X-OVER Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X- Over parameters. Three additional LEDs indicate the status of gain trim (TRIM), polarity (INV) and delay (DLY). A single click with the left mouse but- ton onto this block opens the Output X-Over page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X-Over block within the same project.
	SPEAKER PROCESSING FIR FILTER Block: This block represents the FIR Filter within the corresponding output channel. The graph shows the frequency response that results from the set FIR parameters. The LED indicates if the FIR Filter is being used. A single click with the left mouse button onto this block opens the Output FIR page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding FIR Filter block to any other FIR Filter block within the same project.

	SPEAKER PROCESSING DELAY Block: This displays the SPEAKER PROCESSING DELAY of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate amount of delay being applied. A single click with the left mouse button onto this block opens the Speaker Processing Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other Speaker Delay block within the same project.
	SPEAKER PROCESSING LIMITERS Block: This block provides graphical display of the limiter functions of the corresponding output. The two LEDs indicate whether peak limiter or TEMP limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Limiters block to any other Limiters block within the same project.
	Output Block: A click with the right mouse button onto OUT A or OUT B opens the Copy & Paste menu, which allows copying of all parameters of the corresponding output channel to any other RCM-28 output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied. The numerical field is identical to the numerical field below the level controls in the User panel. The MUTE button is for attenuating the output level of the corresponding output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the output is again active. The MUTE button is virtually disengaged and not lit. The IMP or EXP buttons allow import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for the SPEAKER PROCESSING blocks. The text field allows editing the description of the Speaker Setting file to be exported. Importing a Speaker Setting automatically imports the corresponding Speaker File.

Status Indication

Element	Description
	The MEMORY display shows the number according to the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	NAME indicates the name of the actually audible preset. A new name can be assigned in this field before a preset will be stored to an User Preset.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights green, parameters have been edited and therefore differ from the ones of the preset that is shown.

Store / Store to / Recall

Element	Description
STORE	STORE saves all momentary set DSP parameters into the actually loaded preset.
STORE TO...	STORE TO...saves all momentary set DSP parameters into a selectable User Preset. A click with the left mouse button opens the "Store Preset" dialog, where the preset can be selected. If a new preset name is desired it shall be assigned in the NAME field before pressing the STORE TO ... button. 
RECALL...	RECALL loads and displays all DSP parameters that are stored in the selected preset. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
PROTECT...	The user presets of the RCM-28 can be password protected, so they can only be overwritten when the user knows the password. A click with the left mouse button opens the “Protect User presets” dialog, where the password can be entered. Press the Set Password button to activate the protection. Enter the password and press the Unlock User presets button to deactivate the password protection. 

Preset after Startup

Element	Description
STARTUP PRESET U01	The indicated preset is loaded after power on or reset of the power amplifier. If no preset number is indicated the last used setting is loaded after power on or reset.
ASSIGN ..	Clicking the ASSIGN .. button opens the Set Initial Preset... dialog. In this dialog a factory preset or user preset can be selected as startup pre- set.
X	Clicking the X buttons clears the Startup preset selection.

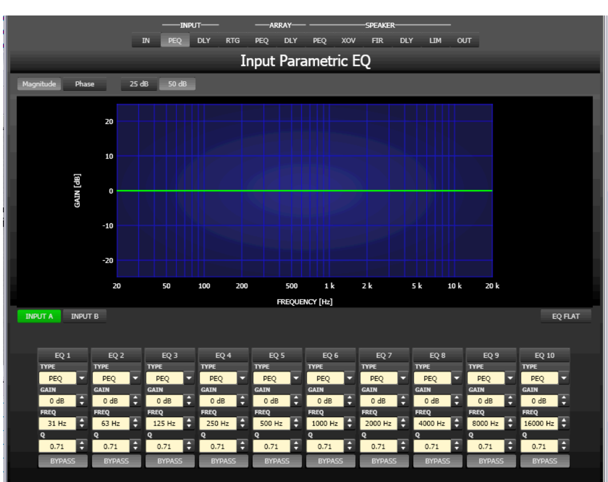
Import / Export of Preset Files

IRIS-Net allows the storing of all DSP parameters of an amplifier together with the according preset name in a file, and to load amplifier parameters from these files. Therefore, IRIS-Net creates a sub-directory \RCM-28 Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory \RCM-28 Presets is permissible.

Element	Description
IMPORT PRESET	After clicking onto IMPORT PRESET an „Open File“ dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a „Save File“ dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store all DSP parameters together with the corresponding file name. „.ds“ is automatically added as file extension.

INPUT PARAMETRIC EQ

Both input channels of the RCM-28 employ 10-band parametric equalizers each, which allow programming highly variable speaker equalization to match a PA-system to different environmental and acoustical requirements.



Graphics Display Indication

Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

Element	Description
INPUT A INPUT B	Switch for selecting input A or input B for filter editing and display. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other PEQ filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency, slope and Q.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	31 / 63 / 125 / 250 / 500 / 1k / 2k / 4k / 8k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lopass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
GAIN 0 dB	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.

ACTIVE			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
EQ FLAT			Press the EQ FLAT button to reset all filters to type PEQ with 0 dB.

Filter Editing via ”Mouse Movement” in the Graphics Display

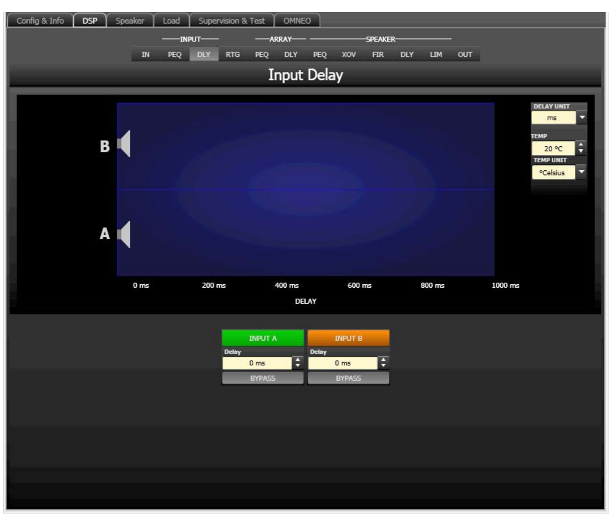
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its magnitude (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs and Hipass/Lopass filters.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

INPUT DELAY

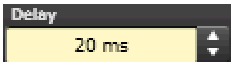
Individual input delays can be set for each input channel of the power amplifier.

HINT: The Input Delay parameter is especially useful for delay lines. In this case the required delay depends only on the position of the delay line and is identical for all band passes, e.g. output channels of the power amplifier. By editing the Input Delay parameter the delays of all output channels routed to this input are adjusted automatically.

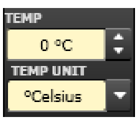


Channel Parameters

Element	Default	Range	Description
INPUT A INPUT B			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding input to any other delay within the same project.

	0 ms	0 to 1000 ms	DELAY allows delaying the corresponding input channel’s audio signal by an adjustable period of time. Entering a value only or a value and unit is possible.
			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the input delay.

General Parameters

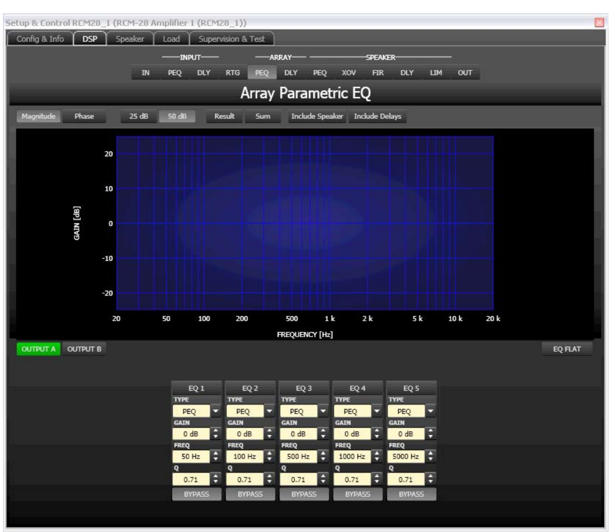
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by “Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

ARRAY PARAMETRIC EQ

All output channels employ 5-band parametric equalizers each, mainly for speaker equalization of arrays. Except for the possibility to select “All pass” as filter type, these filters are identical to the ones of the input EQ's.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Array Parametric EQ (input PEQ), which always provides precise overview and control of the resulting frequency response at this point.

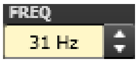
Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level settings and therefore graphically displaying the audible result at the amplifier‘s out- puts.
Sum	The ”Sum” switch causes display of the summed signal of the output channels, including Output level and Mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveal very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally displaying measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
OUTPUT A	Switch for selecting output A or B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other RCM-28 Array EQ-filter bank within the same project.

Filter Parameters

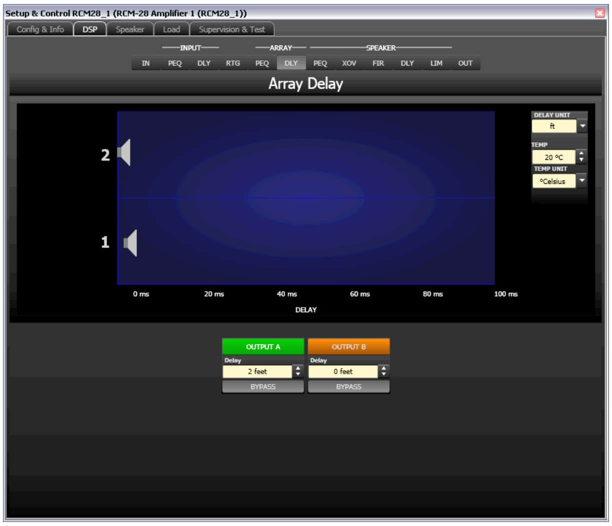
Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency, slope and Q. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.

	50 / 100 / 500 / 1k / 5k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with Allpass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
			Press the EQ FLAT button to reset all filters to type PEQ with 0 dB.

Filter Editing via ”Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

ARRAY DELAY
Individual array delays can be set for each output channel of the power amplifier.

HINT: The Array Delay parameter can be used for adjusting the individual cabinets within a loudspeaker cluster, such as a subwoofer array or a center loudspeaker cluster. For example, in a speaker cluster consisting of two horn-loaded loudspeakers, it is helpful to apply 3-5 ms of delay to one of the loudspeakers in the cluster to improve the coverage in the overlap of the horn patterns. Additionally, the array delay provides a convenient section to apply dedicated delay to individual subwoofer cabinets to create gradient or beam-formed arrays.



Channel Parameters

Element	Default	Range	Description
OUTPUT A			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
Delay 20 ms	0 ms	0 to 100 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
ACTIVE			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the delay.

General Parameters

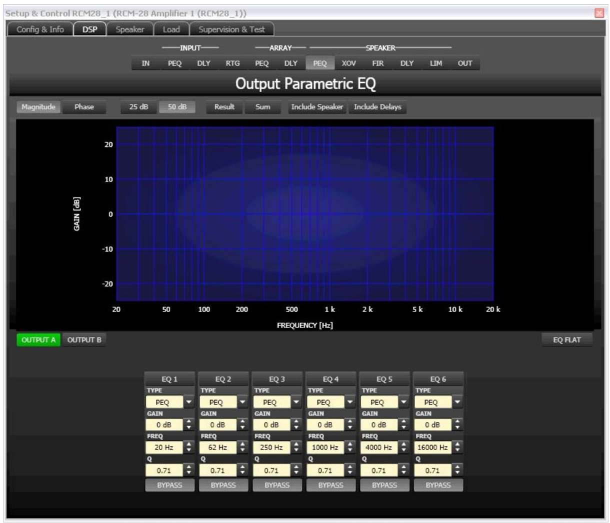
Element	Default	Range	Description
DELAY UNIT ms	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
TEMP 0 °C TEMP UNIT °Celsius	20 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT PARAMETRIC EQ

All output channels employ 6-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select ”All pass” as filter type, these filters are identical to the ones of the input EQ's.



Graphics Display Indication

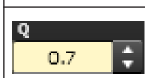
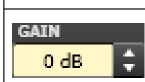
The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Output Parametric EQ, which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB- axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the amplifier‘s outputs.
Sum	The ”Sum” switch causes display of the summed signal of the output channels, including output level and mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveal very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
	Switch for selecting output A or B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other RCM-28 Output EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency, slope and Q. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
	20 / 62 / 250 / 1k / 4k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with All pass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.

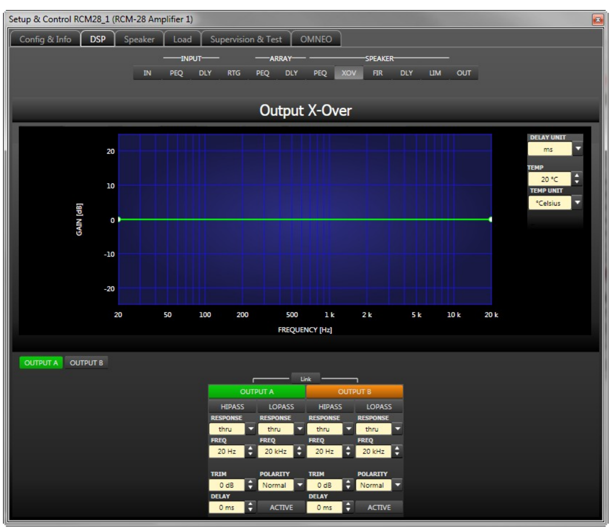
ACTIVE			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
EQ FLAT			Press the EQ FLAT button to reset all filters to type PEQ with 0 dB.

Filter Editing via “Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

OUTPUT X-OVER

The Output X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (e.g. Array Parametric EQ), which always provides precise over-view and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the amplifier‘s outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.

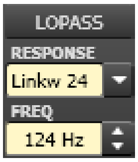
	The ”Sum” switch causes display of the summed signal of the output channels. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.
	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.
	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.

Channel Selection

Element	**Description**
	Switch for selecting output A or B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all X-Over settings of the corresponding output to any other X-Over within the same project.
	Press the Link button to link following parameters of the LOPASS filter of OUTPUT A and HIPASS filter of OUTPUT B: RESPONSE FREQ

Channel parameter

Element	**Default**	**Range**	**Description**
	thru, 20 Hz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI-PASS filters within the same project.

	thru, 20 kHz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.
	0 dB	-30 dB to 6 dB	TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
	Normal	Normal, Inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to “Sum”).
	0 ms	0.0 to 20 ms	DELAY allows the delay of the audio signal from the corresponding output by an adjustable period of time. **HINT: The X-Over Delay parameter is used for the alignment of transducers within cabinets. Optimized delay values are included in Speaker Settings and should not be edited.**
			Press the ACTIVE button to deactivate the delay (Bypass), which allows for quick A / B-evaluation of the actual effect that the delay has on the sound.

Editing X-Over Filters by ”Dragging the Mouse” in the Graphics Display

Active X-Over filters (Response not set to „thru“) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right. A filter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

OUTPUT FIR

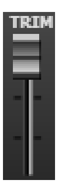
Each output of the RCM-28 offers a 512 taps FIR filter.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the amplifier‘s outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.
Sum	The ”Sum” switch causes display of the summed signal of the output channels. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.

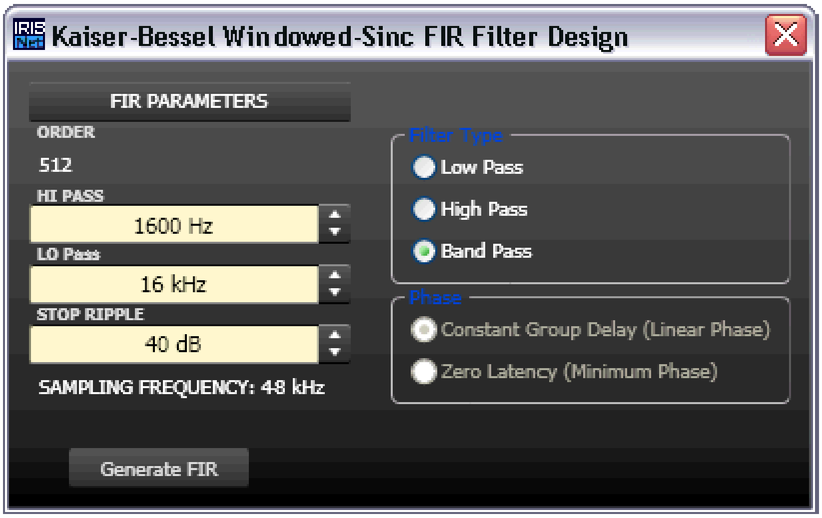
Channel Selection

Element	Description
OUTPUT A	Switch for selecting output A or B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying of all FIR filter settings of the corresponding output to any other FIR filter within the same project.

Channel Parameters

Element	**Description**
	Description of the FIR filter currently in use.
	After clicking onto LOAD the "Open File..." dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
	After clicking on EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click on the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
	Clears the current FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
	Clicking on the NEW button opens the Filter Design dialog.
	Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that the filter has on the sound.
	Adjusts the gain of the signal between -30 dB and +6 dB.
	The fader display shows the numerical value of the current fader setting.
	Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the output is again active. The MUTE button is virtually disengaged and not lit.

FIR-Filter Design

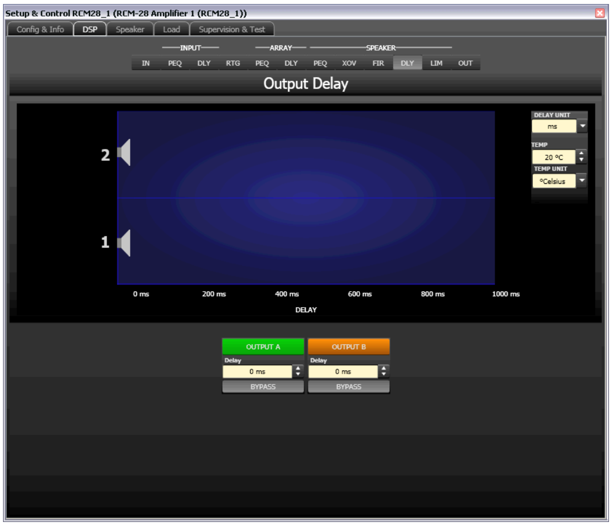


Element	Default	Range	Description
ORDER 512			ORDER of the FIR filter.
HI PASS 1600 Hz	1600 Hz	20 to 19999 Hz	HI PASS sets the cut-off frequency of the Hi pass filter.
LO Pass 16 kHz	16 kHz	21 to 20000 Hz	LO Pass sets the cut-off frequency of the Lo pass filter.
STOP RIPPLE 40 dB	40 dB	21 to 100 dB	STOP RIPPLE sets the slope of the FIR filter.
Filter Type Low Pass High Pass Band Pass	Band Pass		Allows selection of the FIR filter type of the corresponding output channel.
Generate FIR			Press this button to generate the FIR filter.

OUTPUT DELAY

Individual output delays can be set for each output channel of the power amplifier.

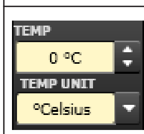
HINT: The power amplifier's output delays can be used to compensate for the positioning of cabinets or speaker arrays relative to each other or the original sound source, for example aligning the PA to the stage or aligning the full-range loudspeakers to the subwoofers. The Output Delay parameter determines the delay time of the corresponding channel or the distance between different loudspeaker clusters.



Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
	0 ms	0 to 1000 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
			Press the ACTIVE button to deactivate the output delay.

General Parameters

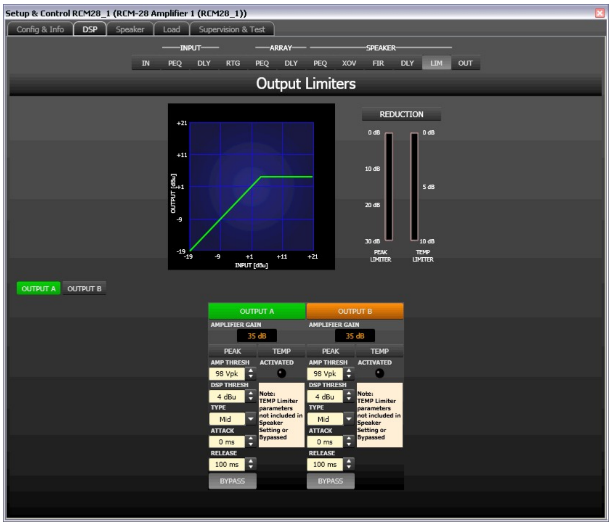
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 ° Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT LIMITERS

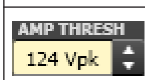
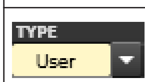
Each output channel of the power amplifier offers a peak limiter and a TEMP limiter. These functions can be accessed via the Output Limiters window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.



Channel Selection

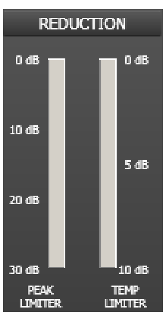
Element	Description
	Switch for selecting output A or B for limiter editing. A click with the right mouse button opens the "Copy & Paste" menu, which allows convenient copying all limiter settings of the corresponding output to any other limiter within the same project.

Limiter Parameters

Element	Default	Range	Description
			Indicates the gain of the power amplifier.
	692 Vpk	22 Vpk to 692 Vpk	AMP THRESH determines the audio signal level at the amplifier output above which the peak limiter starts operating.
	21 dBu	-9 dBu to 21 dBu	DSP THRESH determines the audio signal level at the RCM-28 output above which the peak limiter starts operating.
	User	User, Hi, Mid, Lo, Sub	TYPE allows a user to select a loudspeaker type, and the software will enter appropriate default time constants for the loudspeaker type selected. Speaker settings include factory-defined time constants, so this section is only for use when creating DSP settings from scratch.

	0 ms	0 to 50 ms	ATTACK determines how fast the limiter reduces amplification when the threshold is exceeded.
	100 ms	10 to 1000 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
			Press the ACTIVE button to deactivate the peak limiter.
			The ACTIVATED LED lights green if the TEMP limiter is active. A note is shown below the LED if the TEMP limiter is not included in a set- ting or bypassed.

Gain Reduction Meters

Element	Description
	These indicators show the reduction in dB that is applied to the audio signal by the Peak Anticipation limiter (PEAK) or the TEMP limiter (TEMP LIMITER). Level reduction is indicated as vertical bar graph, the color corresponds to the selected channel.

Editing Limiter Parameters by “Dragging the Mouse” in the Graphics Display

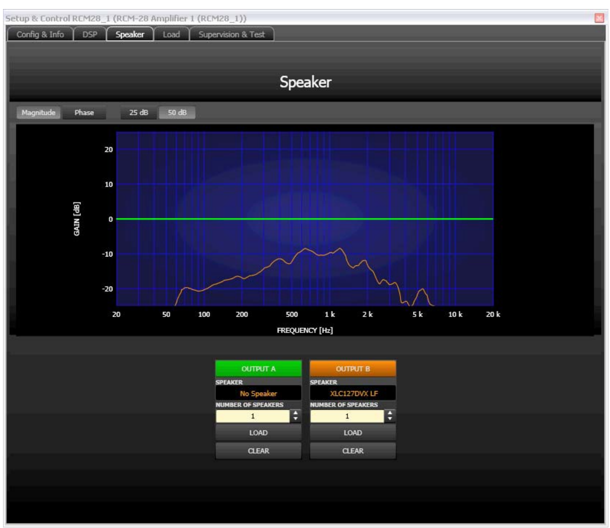
Active limiters (BYPASS button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding limiter by vertically dragging the mouse.

Speaker

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the amplifier channels and display the acoustic results of this virtual combination. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of all common Electro-Voice or DY- NACORD loudspeaker systems. Some examples are provided in the IRIS-Net directory Speaker Files.

HINT: When importing a speaker setting into a output channel the corresponding speaker file is imported automatically.

The speaker data as well as any settings made in this window have no direct influence on the transfer function of the amps. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of amplifier plus speaker transfer functions.
Clicking on the Speaker tab in the Setup & Control window opens the Speaker page.



Indication on the Graphic Display

Element	Description
	Switch for toggling between frequency response (magnitude) and phase response (phase) display
	Switch for adjusting the scale of the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)

Channel Parameters

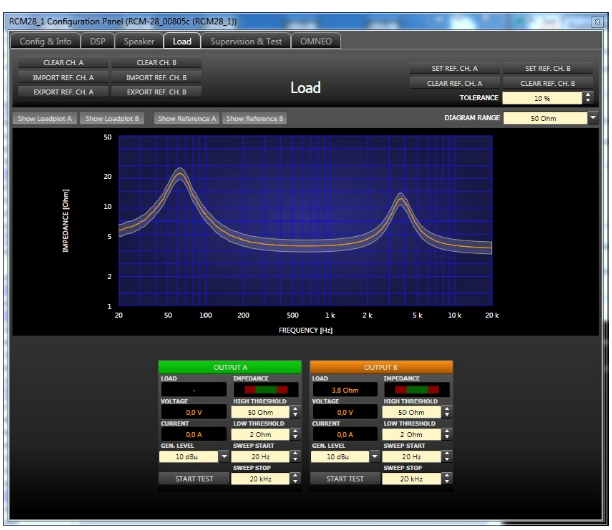
Element	Default	Range	Description
	No speaker		The name of the loaded loudspeaker model is shown in the black-shaded field.

NUMBER 1	1	1 to 10	The NUMBER parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel in the DSP windows.
LOAD			Clicking the LOAD button opens a dialog that allows the selection of the desired speaker file.
CLEAR			Clicking the CLEAR button clears the previously loaded measured speaker data of the selected channel.

Load

The Load window provides access to all settings and functions for testing and monitoring the load connected to the amplifier outputs.
The constantly measured output voltage and output current values of the Remote Power Amplifiers are indicated within the Load window. As soon as the output voltage of the signal present exceeds 150 mV, the resulting load is calculated and indicated. If the set thresholds are being exceeded or fallen short of, a corresponding message appears in the Load display of the Amplifier Control Panel. This dialog box permits to independently set the upper and lower thresholds for each power amp channel.
Within the Load window it is also possible to measure speaker impedance graphs and save them as references. The frequency range (start frequency, stop frequency) and the generator level of the sine-sweep test signal that is generated for this test can be adjusted. Specifying a tolerance field for the saved reference graphs is possible as well. A fault message is displayed in the event that a measurement exceeds or falls short of the tolerance range during system check.

HINT: The speaker impedance test is optimized for low impedance.

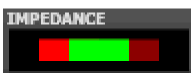
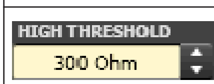


Graphic Display Indication

Element	Default	Range	Description
Show Loadplot A Show Loadplot B			The switches "Show Loadplot A" and "Show Loadplot B" turn the indication of the corresponding impedance graphs ON or OFF.

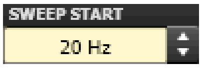
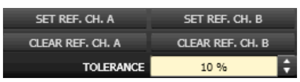
			The switches "Show Reference A" and "Show Reference B" turn the indication of the corresponding reference impedance graphs ON or OFF.
	10 kOhm	50 Ohm...10 kOhm	DIAGRAM RANGE allows zooming in or out the diagram's impedance range (Y- axis).

Parameters And Indications For The Continuous Monitoring Of The Load Connected

Element	Default	Range	Description
			The load display indicates the quotient of measured voltage and current (U/I).
			This indication shows the actual measured load, the progression, and the set value range. The orange needle indicates the actual value. The bright green bar indicates which loads have already been measured while being on-line. A red indication signals that the value exceeded or fell short of the set value range. The dark green area represents the allowable value range for the load of the corresponding power amp channel. The set HIGH THRESH respectively LOW THRESH values define the limits for this value range. Moving the cursor over the indication bar brings up a tool- tip context menu showing the numerical value of the lowest, the highest, and the actually measured load values. Clicking with the right mouse button on the indication bar, followed by a click on Reset, clears the previously measured load values (bright green and red ranges disappear).
			The VOLTAGE display provides continuous indication of the corresponding power amp channel's output voltage.
			The CURRENT display provides continuous indication of the corresponding power amp channel's output current.
	300 Ohm	0 Ohm...10 kOhm	HIGH THRESH sets the upper limit of the allowable impedance range (= minimum load). Once this value is exceeded, an OPEN fault message (line interrupt) appears in the Amplifier Control Panel.
	1 Ohm	0 Ohm... 300 Ohm	LOW THRESH sets the lower limit of the allowable impedance range (= maximum load). Once this value is fallen short of, a SHORTED fault message (line short- circuit) appears in the Amplifier Control Panel.

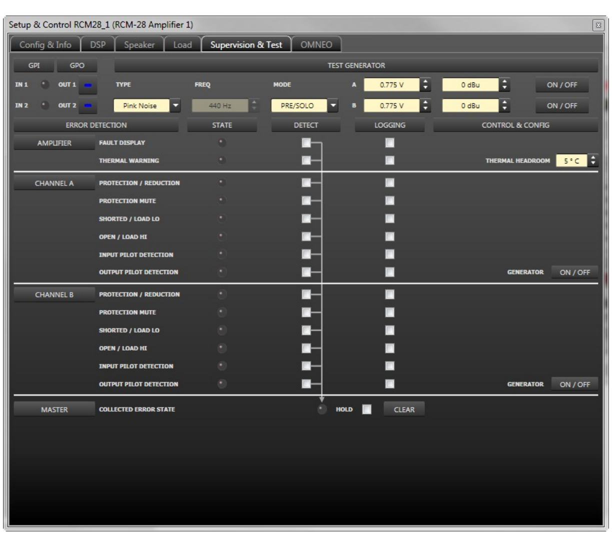
Parameters For Impedance Measurement

Element	Default	Range	Description
	0 dBu	-10 / 0 / 10 dBu	GEN. LEVEL sets the generator level for speaker impedance testing. **CAUTION: Extremely high levels during measurement may result in seriously damaging connected components.**

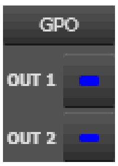
	20 Hz	20 Hz...20 kHz	SWEEP START sets the start frequency of the sine-sweep signal for speaker impedance testing.
	20 kHz	20 Hz...20 kHz	SWEEP STOP sets the stop frequency of the sine-sweep signal for speaker impedance testing.
			Clicking the START TEST soft-key launches the speaker impedance test. The generated sine-sweep signal sweeps over the previously defined frequency range. The graph of the measured impedance values is indicated in the Loadplot display. Clicking this soft-key again cancels the test at any time.
	10%	5%...50%	Clicking the SET REF. CH. A and/or SET REF. CH. B soft-key saves the last test as reference. Clicking the CLEAR REF. CH. A and/or CLEAR REF. CH. B soft-key clears the corresponding reference. TOLERANCE defines the allowable deviation of the impedance graph. Actual measured test results and saved tolerance ranges are being compared during system check. A fault message is displayed if any point of the actual measurement falls outside of this tolerance range. The tolerance range is graphically displayed as spread in the corresponding color that surrounds the reference graph.
			Clicking the CLEAR. CH. soft-key clears the current loadplot. Clicking the EXPORT REF. CH. soft-key stores the current loadplot to a file. Clicking the IMPORT REF. CH. soft-key loades a stored loadplot from a file.

Supervision & Test

The Supervision & Test Dialog integrates functions for testing and monitoring power amplifiers.
You can check control input states and trigger control outputs. A testing generator that provides sine, pink noise and white noise signal output allows acoustical testing. Status indicators for general power amp operation, the two amplifier channels and the load connected, indicate whether everything is okay or where failures occurred. You have the option to choose, which errors are combined and indicated in a general fault message.
A click on the Supervision & Test tab selects the page while in the Setup & Control Window.



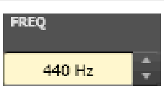
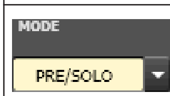
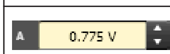
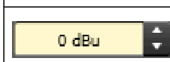
CONTROL INPUTS AND CONTROL OUTPUTS

Element	Description
	This dialog indicates the actual states of the two freely programmable control inputs IN1 and IN2. A green indicator signals "not active", i.e. the control input is open or "high". A red indicator signals "active", in that case the control input is connected to the ground or "low".
	This dialog is for manually controlling the two Open Collector control outputs OUT1 and OUT2. Not engaged (blue) indicates that the control output is deactivated or highly resistive while engaged (red) indicates that the control output is activated and connected to the ground (closed). **HINT: When a control output has already been programmed, the programmed function defines the state of the control output and manual control is not possible.**

For detailed explanation on how to program control inputs and outputs please refer to chapter Config & Info.

TEST GENERATOR PARAMETERS

The test generator allows outputting a selected test-tone at an adjustable level via the power amps channel A and/or channel B which allows testing the cable run from the amplifier output to the connected loudspeaker systems as well as testing the functionality of the loudspeaker components

Element	Default	Range	Description
	Pink Noise	Sine, White Noise, Pink Noise	Type selects the test tone's signal-type
	440 Hz	20...20000 Hz	Freq defines the frequency of the sine signal. This parameter is not available when White Noise or Pink Noise has been chosen as a test-tone signal.
	PRE/SOLO	POST/MIX, POST/SOLO, PRE/MIX, PRE/ SOLO	MIX/SOLO determines whether the generated signal should be mixed with an existing signal. PRE/POST determines if the signal should be generated at the front (PRE) or the rear (POST) of the signal processing chain.
	0.775 V	0.001...2.451 V	These controls are for setting output voltage [V] of the corresponding amplifier outputs.
	0 dBu	-60...+10 dBu	These controls are for setting output level [dBu] of the corresponding amplifier outputs.
	OFF	OFF, ON	These ON/OFF push-buttons activate or deactivate test-tone signal output via corresponding amplifier channels. **CAUTION: Make sure to set a suitable output level, before activating the generator. Extreme output levels can lead to permanent damage of the connected loudspeaker systems!**

Error Detection

Error detection lists the individual STATE of fault indications. Errors collected are amp failure, channel failure, cable interruption, short-circuits, load deviation, ground fault, erroneous communication as well as fault messages of other amps. A green STATE indicator signals normal operation. A red STATE indicator signals error detection.

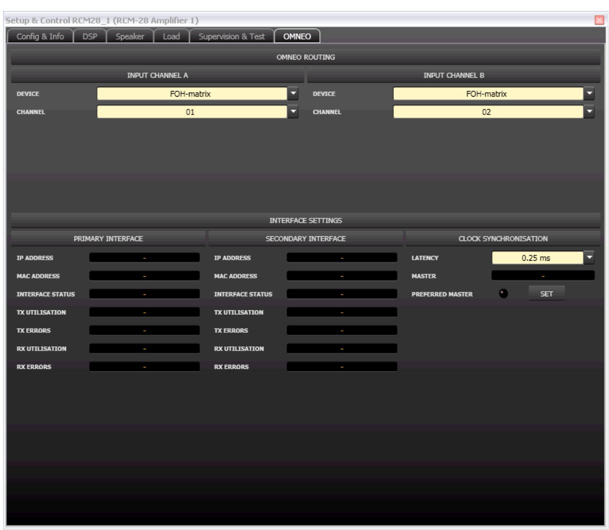
If one of the corresponding DETECT boxes is marked, the state of that message is additionally included in the COLLECTED ERROR STATE. When activating the HOLD option, the indicator stays red after the occurrence of an error. If the HOLD option is not active, indication returns to green, once the fault is no longer detected. Pressing the CLEAR button in the COLLECTED ERROR STATE line resets the indicator from red to green and stored errors are deleted. The COLLECTED ERROR STATE indicator resembles exactly the Amplifier State indicator of the System Check Window. The collected fault state message can be outputted via a control output. For detailed explanation please refer to chapter Config & Info.

Error type	**Description**
FAULT DISPLAY	The red FAULT DISPLAY indicator lights, if a error message is indicated at the amplifier's front LC-Display.
THERMAL WARNING	The power amplifier is protected against thermal overload and will reduce the output power if the internal temperature exceeds a fixed threshold value (please refer to the owner‘s manual for more details). In this case the THERMAL WARNING indicator lights red. With the THERMAL HEADROOM parameter in IRIS-Net the threshold for the THERMAL WARNING indication can be changed to get the warning indication already at lower temperatures, meaning before the amplifier reduces the output power. The THERMAL HEADROOM can be adjusted from 0 to 30 °C which means a temperature warning indication could be configured up to 30 °C before reduction.
PROTECTION / REDUCTION	When the red PROTECTION / REDUCTION indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated. The differentiated protections concept of the power amp results in several protection circuits being activated one after another, which ensures that under normal circumstances the power amplifier will stay in the safe and stabile operating range.
PROTECTION MUTE	When the red PROTECTION MUTE indicator lights, the signal path is switched off. The amplifier needs to be switched off to prevent power amplifier and connected speaker systems from being damaged, this is indicated by the PROTECT and MUTE-LEDs being lit simultaneously.
SHORTED / LOAD LO	This indicator lights red when the measured impedance value of the corresponding power amp output falls below a pre-set minimum or when it is shorted. Setting the minimum value is possible in the Load dialog.
OPEN / LOAD HI	This indicator lights red when the measured impedance value of the corresponding power amp output exceeds a pre-set maximum or when cable interruption is detected. Setting the maximum value is possible in the Load dialog.
INPUT PILOT DETECTION	A remote amplifier's audio inputs support pilot-tone detection and evaluation. Using an externally generated pilot-tone signal allows the monitoring of audio cables and analog input stages. The threshold for 19 kHz pilot-tone evaluation is set to -40 dBu / 7.75 mV. The indicator lights green when an external pilot-tone signal coming from mixer, matrix, controller, etc. is detected. A missing pilot-tone signal or a drop in its level below the evaluation threshold causes the indicator to change to red. Only mark the DETECT box next to the indicator when an external pilot-tone signal actually exists and input monitoring has been configured.

OUTPUT PILOT DETECTION	This indicator is for amplifier monitoring via external pilot-tone signal. In that case, internal pilot-tone generation needs to be switched off to avoid interference between the two signals. Detection and evaluation is performed at the amplifier output. The indicator lights green when a 19 kHz pilot-tone signal with a level of at least -14 dBu / 150mV is detected. A missing pilot-tone signal or a drop in its level below -14 dBu (threshold) results in error detection. The indicator changes to red. **CAUTION: The externally fed pilot-tone signal passes through the entire signal path of the remote amplifier, i.e. the signal is influenced by filtering and x-over settings. When setting the external pilot-tone generator's level, make sure to mind possible amplification/ attenuation applied by internal filters.**
COLLECTED ERROR STATE	COLLECTED ERROR STATE is a collected fault message that combines all error types detected for which the DETECT box had been mar- ked. The HOLD function allows keeping the COLLECTED ERROR STATE for later evaluation while CLEAR clears the indication after remedying the cause of the fault. The COLLECTED ERROR STATE indication is identical to the indication in the Amplifier Status column within the RCM-28 System Check Window.

OMNEO

The OMNEO Dialog integrates functions for configuring and monitoring the OMNEO network. A click on the OMNEO tab selects the page while in the Setup & Control Window.



OMNEO ROUTING

Element	Description
DEVICE	Select the OMNEO device that is the signal source of the audio signal to be used in the RCM-28. Different OMNEO devices can be selected for INPUT CHANNEL A or B.
CHANNEL	Select the OMNEO channel to be used in the INPUT CHANNEL A or B.

INTERFACE SETTINGS

PRIMARY/SECONDARY INTERFACE

Element	Description
IP ADDRESS	Indicates the IP address of the primary or secondary interface.
MAC ADDRESS	Indicates the MAC address of the primary or secondary interface.
INTERFACE STATUS	Indicates the Ethernet speed of the primary or secondary interface (e.g. 1 GBit).
TX UTILISATION	Indicates the current total transmit bandwidth in use.
TX ERRORS	Indicates the number of transmit Cyclic Redundancy Check (CRC) or packet errors detected since last restart.
RX UTILISATION	Indicates the current total receive bandwidth in use.
RX ERRORS	Indicates the number of receive Cyclic Redundancy Check (CRC) or packet errors detected since last restart.

CLOCK SYNCHRONISATION

Element	Description
LATENCY	Select the receive latency time of the RCM-28.
MASTER	Indicates the name of the device that is clock master on that network.
PREFERRED MASTER	The LED lights, if this device is clock master on that network. Pressing the SET button raises the priority of the device in the clock master election. If only one device on the network has this button pressed this ensures that the selected device becomes clock master. When multiple devices have their SET button pressed, the master will be elected from within that group. This is a convenient method of controlling the group of devices from which the master can be selected.

Firmware upgrade

The RCM-28 FWUT FirmWare Update Tool is available in the download sections of www.dynacord.com or www.electro-voice.com

RCM-810

Using RCM-810 remote amplifiers

The IRIS-Net software (Intelligent Remote & Integrated Supervision) runs under Windows and allows configuring, controlling and monitoring a complete PA-system from a single or from several PCs. Any operational status, e.g. power-on, temperature, limiting, activation of protections, deviation of the output impedance, etc., are centrally recorded and displayed, which offers the opportunity to react and interfere even before the occurrence of critical operational states. Programming automated actions that are carried out when exceeding or falling short of certain threshold values is possible as well. All parameters, e.g. power-on/off, mute etc. are controlled in real-time and can be stored in any power amplifier. Monitoring the connected loudspeaker systems is performed by continuously measuring output currents and voltages of individual power amplifier channels. Each exceeding or falling short of set thresholds is instantly signaled and logged. In this way, short-circuits or line interruptions, as they might occur during normal operation, are recognized and displayed immediately.
Furthermore, the RCM-810 provides a control port with freely programmable control inputs and control outputs. Control inputs (GPI's) allow the connection of switches. IRIS-Net offers the possibility to program a variety of logic functions for the inputs. Control outputs (GPO's) allow the connection of external components, which, for example, are used to signal specific states to peripheral equipment. Consequently, an amplifier with a RCM-810 module installed corresponds to highest safety requirements.

Remote Amplifiers

The Remote Control Module RCM-810 can be used in following power amplifiers:

DYNACORD DSA SERIE

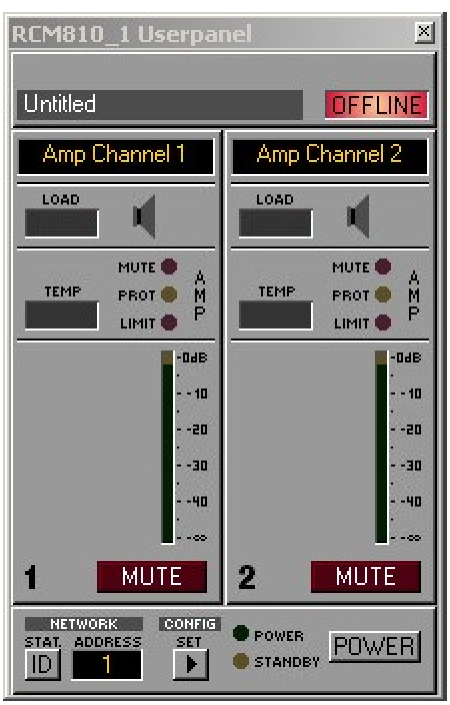
- DSA 8204 2 x 450 W / 4 Ohm or 2 x 650 W / 2 Ohm
- DSA 8206 2 x 600 W / 4 Ohm or 2 x 900 W / 2 Ohm
- DSA 8209 2 x 900 W / 4 Ohm or 2 x 1250 W / 2 Ohm
- DSA 8212 2 x 1200 W / 4 Ohm or 2 x 1800 W / 2 Ohm
- DSA 8405 4 x 500 W / 2...4 Ohm
- DSA 8410 4 x 1000 W / 2...4 Ohm
- DSA 8805 8 x 500 W / 2...4 Ohm

ELECTRO-VOICE CPS SERIE

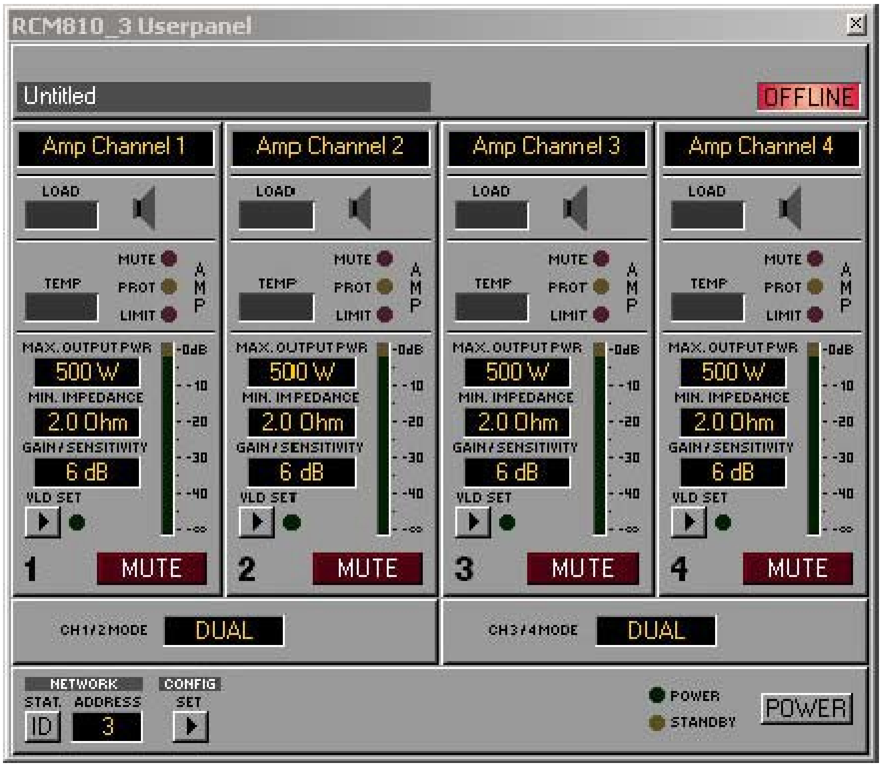
- CPS2.4 2 x 450 W / 4 Ohm or 2 x 650 W / 2 Ohm
- CPS2.6 2 x 600 W / 4 Ohm or 2 x 900 W / 2 Ohm
- CPS2.9 2 x 900 W / 4 Ohm or 2 x 1250 W / 2 Ohm
- CPS2.12 2 x 1200 W / 4 Ohm or 2 x 1800 W / 2 Ohm
- CPS4.5 4 x 500 W / 2...4 Ohm
- CPS4.10 4 x 1000 W / 2...4 Ohm
- CPS8.5 8 x 500 W / 2...4 Ohm

Amplifier Control Panel

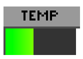
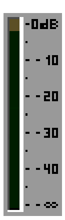
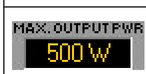
Double clicking with the left mouse button on an amplifier gets you to the Amplifier Control Panel, which provides access to the most important controls and indications of the selected amplifier. Simultaneously opening several Amplifier Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.

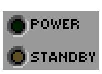


Indications and Functions of the Amplifier Control Panel



Element	Description
☒	Using the left mouse button, click on the Close button to close the Amplifier Control Panel.
Stage Left	A name can be assigned to each amplifier to specify its use or position. Click on the gray-shaded entry field below the Amplifier Type field and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: Entering amplifier names is also possible within the Setup & Control Panel on the Config & Info page. CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
ONLINE OFFLINE	The Online / Offline indicator signals whether the selected amplifier is included in the network or off-line. The red OFFLINE indicator signals that the corresponding amplifier is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding amplifier is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
RCM-810 Output 1	The amplifier channels are named channel 1 to n (n = number of channels). A name can be assigned to each channel to easily identify its allocation and use. Using the left mouse button, click in the entry field and enter the desired name for the channel. Press Return on the keyboard to acknowledge your entry. **HINT: Entering channel names is also possible within the Setup & Control Panel on the Config & Info page.**

	The LOAD indicator shows whether the load connected to the amplifier output is within the allowable range or if short-circuit or line interruption has occurred. The green OK-indication signals that the connected load is between the specified lower and upper limit values. The red OPEN indication signals line interruption. It lights whenever the connected load exceeds the upper limit value. The red SHORTED indication signals short-circuit at the amplifier output. It lights whenever the connected load falls below the lower limit value. **HINT: The connected load is monitored continually as soon as a signal with a voltage of > 150 mV is present at the output. Calculation of signal levels below that threshold is not possible and the indicator shows the last acquired state.**
	The TEMP display shows the amplifier's internal temperature as a graph. The indicator lights green whenever the amplifier is operated in its nor- mal operational temperature range. The indicator lights yellow whenever the amplifier builds up heat because of continuous high output. However, since the internal fans provide sufficient ventilation there is no risk of thermal overload in this state. As soon as temperature indication changes to red, reducing the output level is strongly recommended. Otherwise the amplifier might cease operation because of thermal overload.
	The MUTE indicator lights when the amplifier is muted. This occurs e.g. during speaker switch-on delay.
	When the red PROTECT indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated.
	The LIMIT indicator lights as soon as the internal dynamic limiter is activated, which is the case when the amplifier is operated at maximum out- put. Short-term blinking is not a problem, since the internal limiter controls input levels of up to +20dBu down to a distortion rate of approximately 1%. However, if this indicator lights permanently, reducing the output level is strongly recommended to protect the connected loudspeaker systems from being damaged by capacity overload.
	The Output Level Meters provide indication of the corresponding audio levels at the amplifier outputs. Indication in dB is relative to amplifier full-modulation. A 0dB output level (full-modulation) is indicated in yellow.
 	The MUTE button is for attenuating the output level of the corresponding amplifier output to –∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	MAX. OUTPUT PWR indicates the configured maximum output power of the channel (for 4 and 8 channel amplifiers only).
	MIN. IMPEDANCE indicates the configured minimum impedance of the channel (for 4 and 8 channel amplifiers only).
	GAIN/SENSITIVITY displays the amplifiers constant gain (for 4 and 8 channel amplifiers only).
	Clicking on the VLD SET button of a 4 or 8 channel amplifier opens the VLD tab in the Setup & Control window. The LED indicator next to the button lights when VLD of the channel is activated.

	AMP ROUTING shows how the audio inputs handle the input signals (4 and 8 channel amplifiers only). Possible settings are DUAL and PARALLEL/BRIDGED. Switching the amp routing is only possible locally at the power amplifiers, details can be found in the amplifiers owner's manual.
	Clicking this switch activates the STATUS indicator on the amplifier's rear panel as well as in the amplifier's front panel window in the IRIS-Net software. Normally, the STATUS indicator blinks only during serial communication. Once the STATUS switch is engaged, the STATUS indicator blinks in a steady but fast sequence. This function is meant for checking communication and for identifying or searching an amplifier in a large system setup.
	The address field indicates the set amplifier address. Assigning a new address is also possible by clicking into the field with the left mouse but- ton and entering the desired amplifier address. Available values are 1 to 250. Press Return on the computer keyboard to acknowledge your entry. The assigned address and the address specified by the setting of the selection switch on the amplifier's rear panel have to be identical. Each address can exist only once within a system.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all amplifier-, control and monitoring functions.
	This soft-key allows switching an amplifier on or off. The STANDBY and POWER indicators signal the actual operational status. The Config & Info window allows programming individual power-on delays for each amplifier. **HINT: The power-on delay defaults to "Address * 150 ms". For address 8 the power-on delay default would be for example: 8 * 150 ms = 1200 ms.**
	These indicators show the amp's actual operational status. STANDBY lights whenever the amplifier is in stand-by mode. POWER lights whenever the amplifier is powered-on and ready for operation. If neither one of the indicators lights, the amplifier is either off-line or powered-off.

Setup & Control

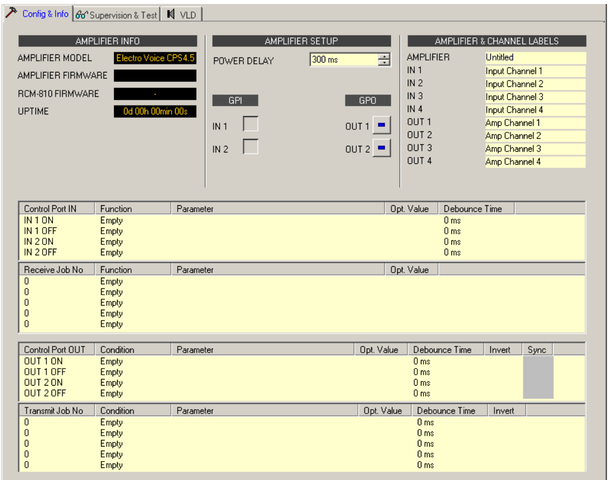
The Setup & Control window allows configuring all amplifier parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

Window	Description
Config & Info	This page provides information about the amplifier and allows making several basic settings as well as programming control functions.
Supervision & Test	This page allows configuring monitoring and surveillance functions and setting the test tone generator.
VLD	This page allows configuring the VLD of 4 and 8 channel amplifiers.

Clicking on the soft key SET in the Amplifier Control Panel opens the Setup & Control window.

Config & Info

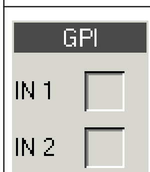
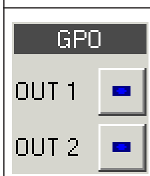
The Config & Info window provides information and basic settings for the selected amplifier. Additionally, editing labels is possible in this window.
To select the page click onto the Configuration & Information register in the Setup & Control Window.



Amplifier Info

Element	Description
AMPLIFIER MODEL	Shows the amplifier type
AMPLIFIER FIRMWARE	Shows the amp software’s version number (operating system, firmware)
RCM-810 FIRMWARE	Shows the remote control module software’s version number (operating system, firmware)
UPTIME	Shows the uptime of the amplifier (standby not included).

Amplifier Setup

Element	Default	Range	Description
POWER DELAY	Address * 150 ms	50...4000 ms 50 ms Steps	Allows programming an amplifier’s power-on delay. Setting different delay times is recommended to prevent the mains fuse from blowing when powering on several amps at the same time. The value defaults to address * 150 ms
			This dialog indicates the actual states of the two freely programmable control inputs IN1 and IN2. A green indicator signals "not active", i.e. the control input is open or "high". A red indicator signals "active", in that case the control input is connected to the ground or "low".
			This dialog is for manually controlling the two Open Collector control outputs OUT1 and OUT2. Not engaged (blue) indicates that the control output is deactivated or highly resistive while engaged (red) indicates that the control output is activated and connected to the ground (closed). **HINT: When a control output has already been programmed, the programmed function defines the state of the control output and manual control is not possible.**

Amplifier & Channel Labels

Element	Description
AMPLIFIER & CHANNEL LABELS AMPLIFIER: Amplifier 1 IN 1: RCM-810 Input 1 IN 2: RCM-810 Input 2 OUT 1: RCM-810 Output 1 OUT 2: RCM-810 Output 2	The labels of an amplifier and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately adopted in the different panels and windows (amplifier control panel, flow diagram, overview). **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

Control Port

A control port offering two control inputs and two control outputs is located on the amplifier's rear panel. The functions of these inputs and outputs can be randomly programmed. For example, the control inputs (GPI) can be used for power- on / stand-by or preset switching as well as for changing parameters. The control outputs (GPO) are for signaling inter- nal statuses. They can directly trigger LEDs, control lamps or relays. In the Supervision & Test window the states of the control inputs are displayed and you have the possibility to switch the control outputs manually. For more information and electrical specifications of the control port, please refer to the amplifier manuals.

Control Inputs: Each status change of a control input can trigger a function. Different functions can be assigned for the opening (OFF) or closing (ON) of a contact.

Example:

Control Port IN	Function	Parameter	Opt. Value	Debounce Time
IN 1 ON	Power	on		0 ms
IN 1 OFF	Power	off		0 ms
IN 2 ON				
IN 2 OFF				

This example shows the programming of two control inputs where IN1 switches the amplifier on or off.

- IN1 ON: Power on (closing the contact of control input 1 switches the amplifier on)
- IN1 OFF: Power off (opening the contact of control input 1 switches the amplifier to stand-by)

Element	Default	Range	Description
Control Port IN		IN 1 ON IN 1 OFF IN 2 ON IN 2 OFF	This provides a listing of the two control inputs and their statuses ON and OFF. The entries in the corresponding lines specify the action when closing (ON) or opening (OFF) a contact.
Function	(empty)		This column allows assigning functions to a control input's statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. Following a status change the assigned function is initiated after the set time interval has past.

Control Outputs: Internal status changes inside of the amplifier, like for example operational faults, alerts when exceeding parameter limits, and internal operational statuses can be signaled to external systems or central control units.
Example:

Control Port OUT	Condition	Parameter	Opt. Value	Debounce Time	Invert	Sync
OUT 1 ON	Power			0 ms		
OUT 1 OFF	Power			0 ms	X	
OUT 2 ON	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms		
OUT 2 OFF	StateFlag	OUTA.THERMPROT,OUTA.PROTECT,OUT...		0 ms	X	

This example shows the programming of two control outputs where OUT1 signals whether the amplifier’s power is switched ON or OFF while OUT2 signals faulty operation.

- OUT1 ON: Power (control output 1 is closed when the amplifier’s power is switched on)
- OUT1 OFF: Invert Power (control output 1 is open when the amplifier’s power is switched off / stand-by mode)
- OUT2 ON: StateFlag (control output 2 is closed when operational faults according to the parameter list have occurred)
- OUT2 OFF: Invert StateFlag (control output 2 is open when no faults have occurred)

Element	**Default**	**Range**	**Description**
Control Port OUT	0	OUT 1 ON OUT 1 OFF OUT 2 ON OUT 2 OFF	This provides a listing of the two control outputs and their statuses ON and OFF. The entries in the corresponding lines specify which status results in the closing (ON) or opening (OFF) of a contact.
Condition	(empty)		This column allows assigning internal events (conditions) to a control output’s statuses. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here you can program delay or debouncing times. An event is signaled following an internal status change and after the specified time interval has past.
Invert	(empty)	(empty) / X	This column allows entering whether a status is signaled when the specified Condition is „true“ (no entry) or „false“ (click „X“ to signal an inverted state).
Sync	(empty)		This column displays the SYNC flag. „X“ specifies that the output is synchronized with a sync-signal. This flag is erased when entering a new Function.

Jobs

For amplifiers to be able to communicate with each other, it is possible to send and receive Job Codes. In principle, a job code is a function number that an amplifier transmits via CAN-bus and that is received and interpreted by another or several other amplifiers. Each amplifier is capable of transmitting and receiving up to 5 different job codes. Programming job codes is nearly identical to the programming of control inputs and outputs.

Receive Jobs: A receive job is a function that is carried out as soon as the corresponding function number (the Receive Job Code) is received.
Example:

Receive Job No	Function	Parameter	Opt. Value
1	Power	on	
2	Power	off	
3	Empty		
4	Empty		
0	Empty		

This example shows the programming of two Receive Jobs. Jobs No. 1 and 2 switch the amplifier’s power on or off
- Receive Job Nr. 1: Power on (receiving Job Code 1 switches the amplifier’s power on)
- Receive Job Nr. 2: Power off (receiving Job Code 2 switches the amplifier into stand-by mode)

Element	Default	Range	Description
Receive Job No	0	1...1023	Here, you can specify which incoming job code numbers a specific amplifier recognizes. Entering random numbers between 0 and 1023 is possible.
Function	(empty)		This column allows assigning an individual function to each job code received. Clicking the desired line in the Function menu opens a dialog field that shows all accessible functions. The table "Input and Receive Job Functions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Input and Receive Job Functions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.

HINT: Programming identical control functions or receive jobs for several amps is easily accomplished by creating a group that includes all the desired amps and afterwards perform the programming in the group’s Configuration & Information dialog. All settings are automatically applied to all amplifiers of that group, which saves time and effort and additionally reduces the risk of programming errors.

Transmit Jobs: Transmit Job defines a function number that is sent as soon as a specific internal event (condition) occurs in the amplifier.
Example:

Transmit Job No	Condition	Parameter	Opt. Value	Debounce Time	Invert
1	GPI	IN1		0 ms	
2	GPI	IN1		0 ms	X
3	GPI	IN2		0 ms	
4	GPI	IN2		0 ms	X
0	Empty			0 ms	

This example shows the programming of four Transmit Jobs. Jobs No. 1 and 2 are triggered by control input 1. Jobs No. 3 and 4 are triggered by the status signaled from control input 2. The fifth transmit job has not been configured.
- Transmit Job Nr. 1: GPI IN1 (Job Code 1 is transmitted when control input 1 is closing)
- Transmit Job Nr. 2: Invert GPI IN1 (Job Code 2 is transmitted when control input 1 opens)
- Transmit Job Nr. 3: GPI IN2 (Job Code 3 is transmitted when control input 2 is closing)
- Transmit Job Nr. 4: Invert GPI IN2 (Job Code 4 is transmitted when control input 2 opens)

Element	Default	Range	Description
Transmit Job No	0	1...65536	Here, you can specify which job code numbers an amplifier transmits on the occurrence of specific events. Entering random numbers between 0 and 65536 is possible.
Condition	(empty)		This column allows specifying an event (condition) that triggers the corresponding transmit job code. Clicking the desired line in the Condition menu opens a dialog field that shows all accessible functions. The table "Output and Transmit Job Conditions" lists all functions together with their individual settings.
Parameter	(empty)		Here you can set the different function parameters. For more information, please refer to the table "Output and Transmit Job Conditions".
Opt. Value	(empty)		Certain functions allow specifying optional parameter values.
Debounce Time	0 ms	0...10027 ms 16.33 ms Steps	Here, you can program delay or debouncing times. A transmit job code is sent following a specific event and after the specified time interval has past.
Invert			This column allows entering whether a job code is transmitted when the specified Condition is "tru" (no entry) or "false" (click "X" to signal an inverted state).

Input and Receive Job Function: The following table lists all functions together with their individual settings, which can be triggered via control input or Receive Job.

Function	Parameter	Opt. Value	Function executed
Empty	-	-	None
Power	off on flip		Power Off (Standby) Power On Power-status change (ON to Stand-by and reverse)
Absolute	Mute	0 = not muted, 1 = muted	Set the specified absolute parameter value for the selected parameter
Toggle	Mute		Changes the status of the selected parameter
Memoflag	Set, Clear, Toggle Memoflags 1 - 16		Sets, erases or changes selected memory flags. Up to 16 memory flags are available and simultaneously accessible.

Output and Transmit Job Conditions: The following table lists all amplifier statuses that can be used for triggering control outputs or for sending Transmit Job Codes.

Function	Parameter	Opt.Value	Invert	Triggering Event / Status Change
Empty	-	-		Not configured

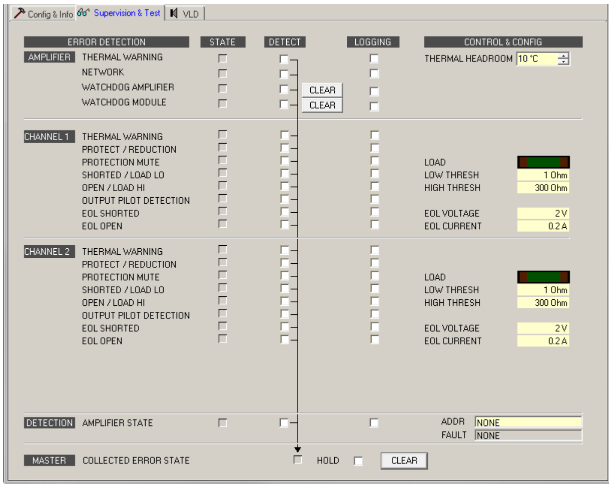
Power			X	Power On Power Off (Standby)
Absolute	Mute	0 = not muted, 1 = muted	X	Set parameter value reached or exceeded Set parameter value declined
Temp	Temperature in °C		X	Set temperature reached or exceeded Set temperature declined
VU	Out 1...n	Level in dB	X	Set level reached or exceeded Set level declined
GPI	IN 1, IN 2		X	Control input 1 / 2 closed (ON) Control input 1 / 2 open (OFF)
Stateflag	All internal fault conditions		X	Single or several stateflags set None of the selected stateflags set
Memoflag	Enable for selected flags as well as bit-pattern of flags 1 - 16		X	Memory flags match the selected bit-pattern Memory flags do not match the selected bit-pattern

Supervision & Test

The Supervision & Test Dialog integrates functions for testing and monitoring power amps. Status indicators for general power amp operation, the amplifier channels and the load connected, indicate whether everything is okay or where failures occurred. You have the option to choose, which errors are combined and indicated in a general fault message.
A click on the Supervision & Test tab selects the page while in the Setup & Control Window.

Error Detection

Error detection lists the individual STATE of fault indications. Errors collected are amp failure, channel failure, cable interruption, short-circuits, load deviation, ground fault, erroneous communication via the CAN bus as well as fault messages of other amps. A green STATE indicator signals normal operation. A red STATE indicator signals error detection.
If one of the corresponding DETECT boxes is marked, the state of that message is additionally included in the COLLECTED ERROR STATE. When activating the HOLD option, the indicator stays red after the occurrence of an error. If the HOLD option is not active, indication returns to green, once the fault is not detected anymore. Pressing the CLEAR button in the COLLECTED ERROR STATE line resets the indicator from red to green and stored errors are deleted. The COLLECTED ERROR STATE indicator resembles exactly the Amplifier State indicator of the System Check Window. The collected fault state message can be outputted via a control output. For detailed explanation please refer to chapter Config & Info.

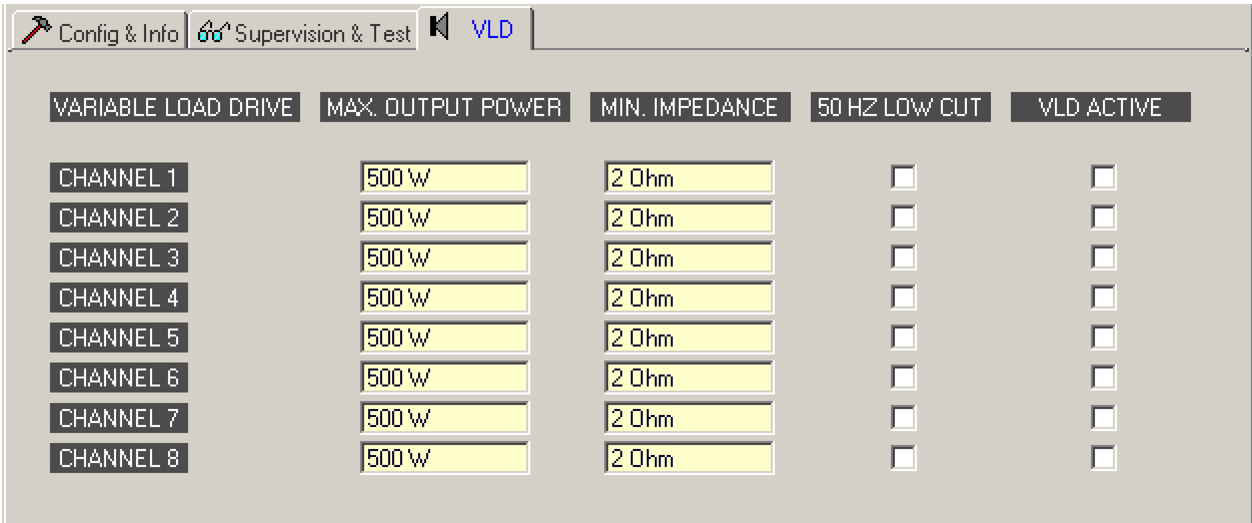


Element	Description
AMPLIFIER	
THERMAL WARNING	This indicator lights red when the power amp's temperature exceeds the pre-set threshold, which defaults to 110 °C. If necessary, changing the temperature threshold is possible via THERMAL LIMIT. The power amp, however, is protected against thermal overload at all times, independent of the indication.
NETWORK	This indicator shows whether communication at the CAN bus interface is normal (green) or when a problem exists (red). The power amp automatically detects whether commands from a PC or another central control unit are missing and signals the problem via the Communication Flag.
WATCHDOG AMPLIFIER	This indicator lights red when the power amp‘s watchdog has been activated. Press the CLEAR button to clear the indication.
WATCHDOG MODULE	This indicator lights red when the module‘s watchdog has been activated. Press the CLEAR button to clear the indication.
CHANNEL 1...n	
THERMAL WARNING	This indicator lights red when the power amp's output channel temperature exceeds the pre-set threshold. The power amp, however, is protected against thermal overload at all times, independent of the indication.
PROTECT / REDUCTION	When the red PROTECT / REDUCTION indicator lights, one of the internal protections (thermal overload, short-circuit, Back-EMF, HF at the output, etc.) has been activated.
PROTECTION MUTE	When the red PROTECT MUTE indicator lights, the signal path gets switched off. The amplifier needs to be switched off to prevent power amplifier and connected speaker systems from being damaged, this is indicated by the PROTECT and MUTE-LEDs being lit simultaneously.

LOAD	This indication shows the actual measured load, the progression, and the set value range. The orange needle indicates the actual value. The bright green bar indicates which loads have already been measured while being on-line. A red indication signals that the value exceeded or fell short of the set value range. The dark green area represents the allowable value range for the load of the corresponding power amp channel. The set HIGH THRESH respectively LOW THRESH values define the limits for this value range. Moving the cursor over the indication bar brings up a tool-tip context menu showing the numerical value of the lowest, the highest, and the actually measured load values. Clicking with the right mouse button on the indication bar, followed by a click on Reset, clears the previously measured load values (bright green and red ranges disappear).
SHORTED / LOAD LO	This indicator lights red when the measured impedance value of the corresponding power amp output falls below a pre-set minimum LOW THRESH or when it is shorted.
OPEN / LOAD HI	This indicator lights red when the measured impedance value of the corresponding power amp output exceeds a pre-set maximum HIGH THRESH or when cable interruption is detected.
OUTPUT PILOT DETECTION	This indicator is for amplifier monitoring via external pilot-tone signal. Detection and evaluation is performed at the amplifier output. The indicator lights green when a 19 kHz pilot-tone signal with a level of at least -14 dBu / 150mV is detected. A missing pilot-tone signal or a drop in its level below -14 dBu (threshold) results in error detection. The indicator changes to red.
EOL SHORTED	This indicator lights red if the voltage at the amplifier‘s output is below the EOL VOLTAGE threshold.
EOL OPEN	This indicator lights red if the current at the amplifier‘s output is below the EOL CURRENT threshold
EOL VOLTAGE	Threshold for EOL SHORTED error indication.
EOL CURRENT	Threshold for EOL OPEN error indication.
DETECTION	
AMPLIFIER STATE	A RCM-810 remote amplifier is capable of detecting and indicating the operational state of other RCM-810 amps within a CAN network. The addresses of all amps that are to be monitored are entered in the ADDR field, e.g. 2-4,6,11. The FAULT field indicates the amp addresses for which errors have been detected and the COLLECTED ERROR STATE has been activated (red). The indicator changes to red as soon as at least one amplifier in the list shows erroneous operation.
MASTER	
COLLECTED ERROR STATE	COLLECTED ERROR STATE is a collected fault message that combines all error types detected for which the DETECT box had been mar- ked. The HOLD function allows keeping the COLLECTED ERROR STATE for later evaluation while CLEAR clears the indication after remedying the cause of the fault. The COLLECTED ERROR STATE indication is identical to the indication in the Amplifier Status column within the RCM-26 System Check Window.

Variable Load Drive (VLD)

The VLD window allows configuration of the "Variable Load Drive" mode of the amplifier output channels. For each channel the maximum output power and minimum impedance can be set. Additionally a 50 Hz Low Cut Filter can be activated for each channel. To select the page click onto the VLD register in the Setup & Control Window.



Element	Description
MAX. OUTPUT POWER	Set the maximum output power of the channel.
MIN. IMPEDANCE	Set the minimum impedance connected to the channel.
50 HZ LOW CUT	Activates the 50 Hz Low Cut filter.
VLD ACTIVE	Activates the VLD mode of the channel. This checkbox has an effect only if the amplifier output channel is operated in output mode 2 Ohm/ VLD. In the output modes 4 Ohm, 70 V or 100 V the setting of the checkbox has no effect on the output channel.

Firmware Upgrade

The firmware of RCM-810 remote amps is stored in a FLASH-memory chip. This technology has been chosen to be able to provide the users with new software without the hassle of physically exchanging memory chips inside of a remote amplifier. Using IRIS-Net, upgrading the firmware is possible via the CAN Remote Control Interface. In this way you can install new firmware and future software extensions to always keep your Remote Amplifier System up-to-date.
The RCM-810 firmware is divided into a part for basic amplifier functions (e.g. power on/off, CAN communication) and a part for extended functions (e.g. signal processing). Even if the firmware update procedure does not finish successfully, basic amplifier functions stay functional and the update procedure can be repeated.

Caution!
Upgrading the firmware is always a very sensible procedure – comparable to updating the OS in the FLASH-memory of a PC. Therefore, obeying the following precautions and instructions is absolutely mandatory:
Consequences

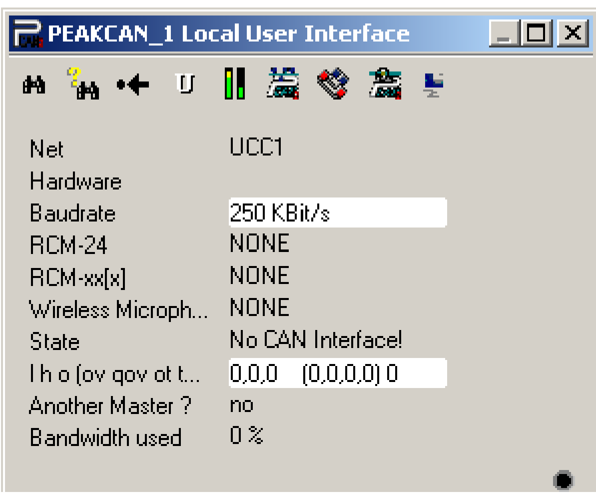
1. Simultaneously upgrading the firmware of more than four remote amplifiers is not recommended.
2. Only connect the remote amps to the CAN Remote Control network that are to be updated. Disconnect any other remote amps from the CAN-bus during the upgrade. Make sure to carefully mind all regulations for the CAN Remote Control network, especially the 120 Ω termination at both ends of the bus.

HOW TO UPGRADE THE FIRMWARE

Necessary preparations

1. Connect the desired remote amp(s) via CAN-bus to your PC.
2. Start the IRIS-Net software and open your project. Your remote amps and the icon of a PC with CAN-label should appear on your screen. The PC-icon represents the CAN-interface of your PC or notebook.

3. Double-clicking onto the PC-icon opens the CAN-interface window. CAN-bus status and connected remote amps are displayed. This window display is available in off-line mode.



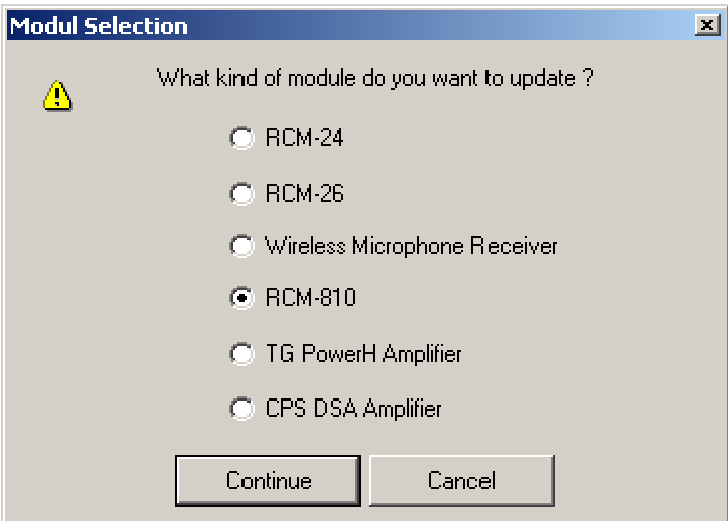
Make sure to check the following parameters before upgrading:

Element	Description
Baud rate	Indicates the set baud rate. Normally you don't have to change the system's baud rate for upgrading.
RCM-xx[x]	Indicates the addresses of the remote amps connected. Make sure that the addresses shown are only the ones of remote amps that you want to upgrade.
State	Indicates the CAN-interface status. This has to read "OK". Otherwise, starting the firmware upgrade is not permissible.
I h o (...	Indicates different error flags. Under no circumstance the first 3 digits may rise. Clicking into the white field and entering "0" resets the error flags.
Bandwidth used	Indicates the used bandwidth of the CAN-bus in percent. Make sure to check that the CAN-bus is not too busy, i.e. high data traffic.

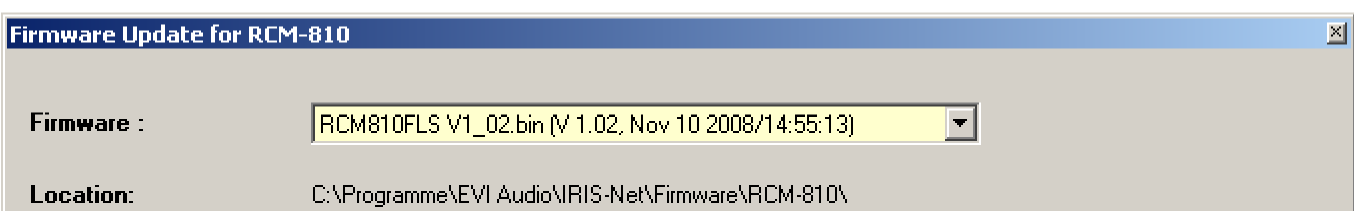
You can update the firmware of the remote control module or the firmware of the amplifier itself.

Firmware update of the Remote Control Module

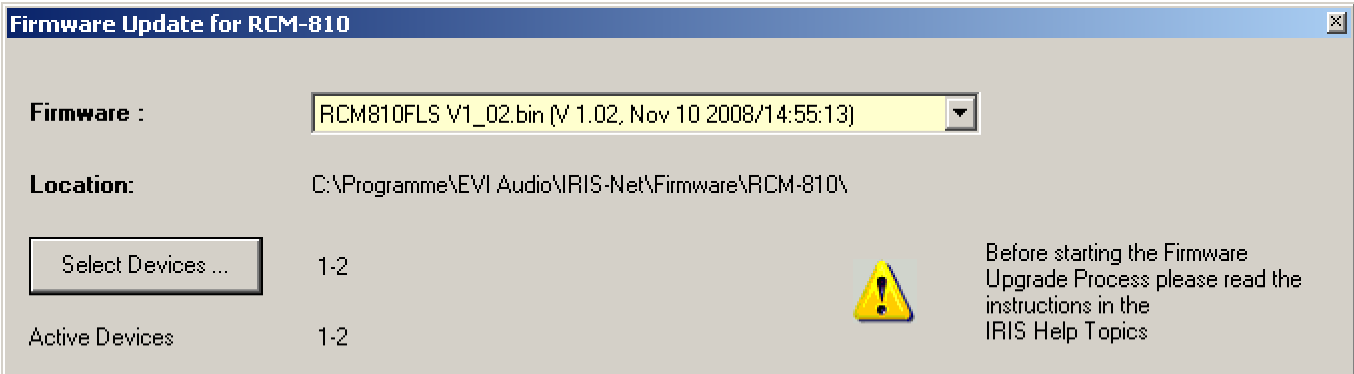
1. The CAN-interface window provides a toolbar (top line). Clicking onto the U-icon (Update) opens an Module Selection dialog.
2. Select RCM-810 and click onto "Continue".



3. The actual firmware file including version number and date is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date remote amplifier firmware version. The corresponding file is located in the directory: \IRIS-Net\Firmware\RCM-810. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.



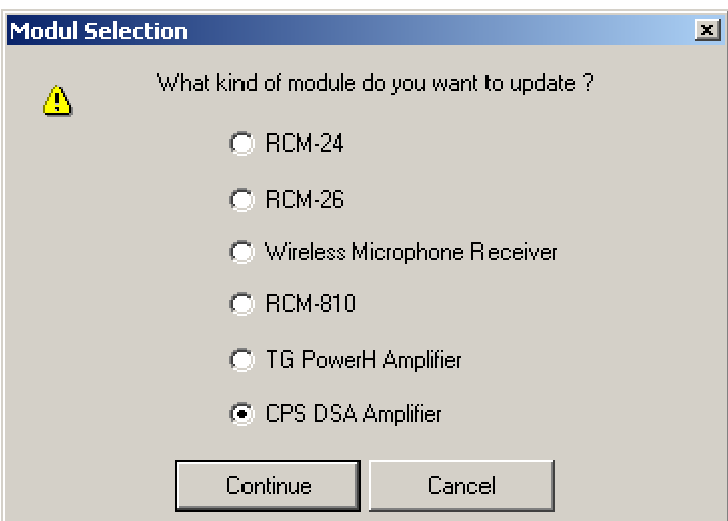
4. Click onto the button “Select Devices...” to open a list of all remote amps connected. Select the amp(s) that you want to update and click the “OK” button. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus. When performing the firmware upgrade for the first time, connecting only a single amplifier is recommended to become familiar with the upgrading procedure.
5. Click onto the button “Select Devices...” to open a list of all remote amps connected. Select the amp(s) that you want to update and click the “OK” button. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus. When performing the firmware upgrade for the first time, connecting only a single amplifier is recommended to become familiar with the upgrading procedure. The addresses of the selected remote amp(s) are shown in the firmware update window on the right side next to the button “Select Devices...” and in the line “Active Devices”.
6. Clicking onto “Start Update” starts the upgrade procedure. The single steps of the update are shown in the “Messages” window. The progress of some parts of the upgrade which take a little longer is indicated through dots behind the corresponding name. The message “ok” has to appear at the end of each line. The following example shows how to upgrade the firmware of the remote amplifiers with the addresses 1 and 2 to firmware version V 1.02.



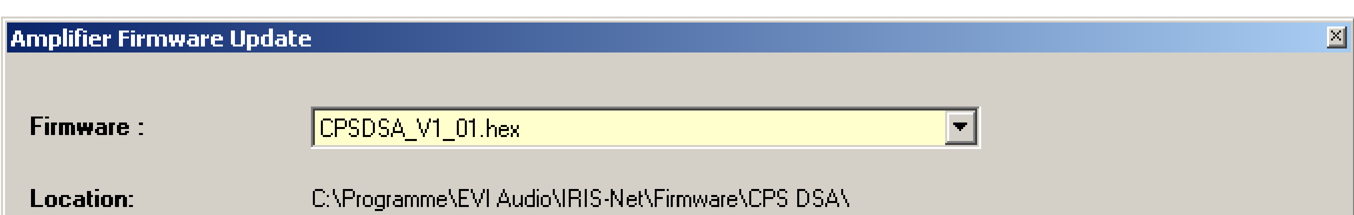
7. The message “Finishing .. ok” indicates that upgrading has been successful. The remote amp(s) are reset. Afterwards they are again ready for operation. The upgrade procedure is finished and you can close the dialog window or proceed with upgrading other remote amps.

Firmware update of the amplifier

1. The CAN-interface window provides a toolbar (top line). Clicking onto the U-icon (Update) opens a Module Selection dialog.
2. Select CPS DSA Amplifier and click onto Continue.

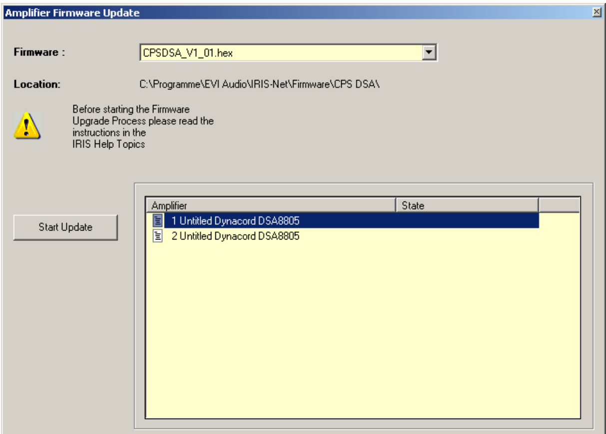


3. The actual firmware file including version number and date is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date remote amplifier firmware version. The corresponding file is located in the directory: \IRIS\Firmware\CPS DSA. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.



4. Select the amplifier that you want to update and click the “OK” button. The list should only show amps that you want to update. No other amplifier should be connected to the CAN-bus.

5. Clicking onto “Start Update” starts the upgrade procedure. The single steps of the update are shown in the “Messages” window. The progress of some parts of the upgrade which take a little longer is indicated through dots behind the corresponding name. The message “ok” has to appear at the end of each line. The following example shows how to upgrade the firmware of the remote amp with the address 1 to firmware version V 1.01.



6. The message “Finishing .. ok” indicates that upgrading has been successful. The remote amp(s) are reset. Afterwards they are again ready for operation. The upgrade procedure is finished and you can close the dialog window or proceed with upgrading other remote amps.

ADDITIONAL NOTES CONCERNING A FIRMWARE UPGRADE

– The line “Active Devices” indicates which of the selected remote amps are still to be updated. Amps for which the update process timed out are taken off the list. These devices are still capable of receiving upgrade commands. However, the software does not wait for acknowledgements of the concerned amps any longer.
– If the IRIS-Net-software recognizes an error or “Time Out” during upgrading, it automatically switches to “Single Step” mode, which offers the possibility to repeat the upgrade in single steps. If a “Time Out” message is displayed while upgrading is in progress, under no circumstance switch off any amps!
– As soon as “Single Step” is checked off, all buttons below the single step field become active. The upgrade can now be performed manually, step- by-step in the sequence as described below. If one of the commands does not finish “ok”, you have to restart the upgrade procedure from the beginning.

Step	**Description**
Start Update	Activates update mode for the selected devices. The messages window shows “Update started (addresses)” .. and after a short period of time “ok”.
Verify	Compares the firmware installed in the remote amps with the selected firmware file. The messages window shows “Verifying (addresses)”...a progression-bar indicates the approximate duration of the process. Detected differences are indicated at the end of the process, e.g. “done, Errors detected for...”. If no errors time-outs are detected, you can proceed with the update.
Erase Flashes	Deletes the actual firmware and clears the FLASH-memory of a remote amplifier. The messages window shows “Erasing (addresses)”..... and after a short period of time “ok”.

Program	Loads the new firmware into the FLASH-memory of a remote amplifier. The messages window shows “Programming (addresses)"........ A progression-bar indicates the approximate duration of the programming. “ok” appears in the message window after some time.
Checksum	Evaluates the checksum of the newly installed firmware. The messages window shows “Checksum (addresses)” .. and after a short period of time “ok”. This is a short form of the “Verify” process.
Stop Update	Finishes upgrading. The messages window shows “Finishing (addresses)" .. and after a short period of time “ok”. The remote amps quit the update mode and start in nor- mal mode. Now, you can exit the upgrade dialog or proceed with upgrading other remote amps.

– If “Time Out” errors still occur during the programming, repeat the procedure in single step mode in the following sequence: Start Update - Program.
– If the checksum evaluation shows errors, repeat the entire upgrade procedure. Don’t forget to uncheck “Single Step” mode, for the upgrade to run automatically.

REV WIRELESS MICROPHONE SYSTEM

Introduction

The REV Wireless Microphone system combines frequency agility and ease of use like no other. The REV transmitters and receivers operate over a 24 MHz bandwidth in the UHF portion of the spectrum. The high quality audio circuitry and advanced Radio Frequency (RF) signal processing offer broadcast quality signal-to-noise and audio clarity.
The system features include:
Advanced ClearScan technology for selecting clear channels and inter-modulation free groups

- 960 Radio Channels, user programmable or factory installed
- LCD Displays for ease of viewing
- Patented DSP Phase Diversity System
- CAN Bus Port for computer monitor, control, and updating (connection via UCC1 USB-CAN Converter)
- Adjustable Unbalanced Line Level 1/4 inch output jack
- Adjustable Balanced Microphone Line Level XLR output jack
- Front Panel Power ON/OFF Switch
- Permanent Flash Memory for frequency/system storage
- Front Panel Software Control of Squelch and Audio Output settings
- Combination Squelch (Amplitude and Tone) system prevents false squelch
- Lockout feature to prevent accidental channel changes
- Sound Check mode to speed walk testing and provide tangible results
- Backlit LCD displays on transmitters for easy adjustment on dark stages
- "Smart" battery feature in the 9V transmitters means there is no wrong orientation
- Interchangeable heads on the handheld transmitters
- Cast magnesium case on the bodypack transmitter

– Optimized Guitar settings on transmitter and receiver

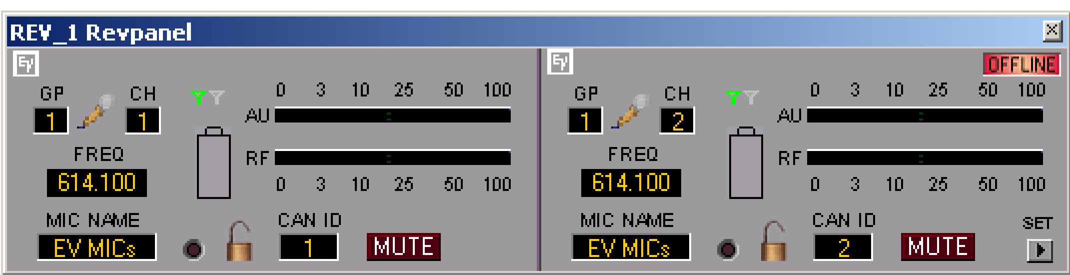
The REV Wireless Microphone system can be connected to the PC using one of following devices:
– UCC1 USB-CAN Converter
– Electro-Voice NetMax N8000 System Controller
– DYNACORD P 64 Matrix Manager

REV Device

The REV Devices can be accessed from the Objects Bar (listed under the category Electro-Voice) or from the separate Devices window, which opens after clicking on the item Add Device. Add Device is available from the IRIS-Net Configuration menu or from the contextual menu within the IRIS-Net worksheet.
To add devices to an IRIS-Net project, first select the desired device REVS or REVD in the Object Bar (or from the Devices window) and then drag and drop it into the worksheet. A dialog box opens, which lets you specify device-related settings such as amount of desired devices, address range, and interfaces.

REV Control Panel

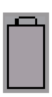
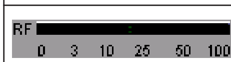
Double clicking with the left mouse button on an REV device gets you to the REV Control Panel, which provides access to the most important controls and indications of the selected microphone system. Simultaneously opening several REV Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.



Caution!
REV Control Panel will be displayed with a red title bar and a text stating “Low Battery” when the cur- rent battery level goes below the current Low Battery Alarm Level. Also REV Control Panel is popped up if it is not already open or brought to the foreground if it is already open.
Consequences

Indications and Functions of the REV Control Panel

Element	Description
	Using the left mouse button, click on the Close button to close the REV Control Panel.
ONLINE OFFLINE	The Online / Offline indicator signals whether the selected receiver is included in the network or off-line. The red OFFLINE indicator signals that the corresponding receiver is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding receiver is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately trans- mitted and active.

	The receiver can be assigned a group. Click on the gray-shaded entry field below the GP label and enter the desired group. Press Return on the keyboard to acknowledge the entered group. The group value can only be in the range 1-21.
	The REV system guitar optimization mode requires both the transmitter and receiver be placed into guitar mode. Clicking on the micro- phone bitmap puts the device in guitar mode and bitmap changes to a guitar. Clicking on the guitar bitmap puts the device in voice mode and bitmap changes to a microphone.
	The receiver can be assigned a channel. Click on the gray-shaded entry field below the CH label and enter the desired channel. Press Return on the keyboard to acknowledge the entered channel. The channel value can only be in the range 1-16.
	The FREQ indicator displays the current operating frequency of the receiver. The change to the frequency value is accepted only when online and only for a user group. The frequency should be within the operating frequency range. Press Return on the keyboard to acknowledge the entered frequency
	A name can be assigned to each microphone to specify its use or position. Click on the gray-shaded entry field below the MIC NAME label and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: Name shall be alphanumeric and no special character except space is allowed. Name shall be utmost 8 characters in length.**
	The antenna diversity indicator signals the current diversity status. The green indication on the left antenna indicates that the signal is fed through the left antenna. The green indication on the right antenna indicates that the signal is fed through the right antenna.
	The multicolor battery level indicator shows the current battery status of the body-pack or the microphone. The indication is active only when the group and channel and the operating frequency on the REV receiver and microphone are same. The threshold battery level below which the status is indicated in red depends on the low battery alarm level property configurable on the REV Configuration dialog. A green indicates the battery is full.
	The AU display shows the current level of audio input to the receiver. The value in the range 0-10 is indicated in green, 11-50 is indicated in yellow and 51-100 is indicated in red
	The RF display shows the current strength of the RF signal input to the receiver. The value in the range 0-10 is indicated in red, 11-50 is indicated in yellow and 51-100 is indicated in green.
	Error status led is a priority based indicator that displays error status in battery level (High priority), audio level (Medium priority) or RF signal strength (low priority). The led flashes in red for a battery error, orange for an audio level error, amber for RF signal strength error based on the precedence of their priority levels. The led is hollow if there are no errors.
	The edition of REV properties can be locked or unlocked. Pressing on the lock bitmap on the panel locks the edition. Pressing on lock again unlocks it. **HINT: Locking or unlocking can be done only when no operation is performed currently on the receiver.**

	CAN ID allows you to modify the address of the receiver on the worksheet. Click on the gray-shaded entry field below the CAN ID label and enter the desired CAN address. Press Return on the keyboard to confirm the CAN address entered. This address must be the same as the CAN address specified in the REV menu. Each address can only be used once within a system. Both the REV-S (single receiver) and the REV-D (dual receivers) use two consecutive CAN addresses. The CAN address of the left receiver (REV-D only) is always odd, the CAN address of the right receiver (both REV-S and REV-D) is always one higher than the left address and hence it is always even. In the IRIS-Net worksheet only the even CAN address is displayed on the receiver icon. **HINT: Changing the address of the left receiver will automatically change the address of the right receiver and vice-versa.**
 	The MUTE button is for attenuating the microphone level to $-\infty$. Clicking the MUTE button with the left mouse button mutes the micro- phone. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the microphone is again active. The MUTE button is virtually disengaged and not lit.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all microphone-, control and monitoring functions.

Setup & Control

The Setup & Control window allows configuring all microphone parameters. It also provides access to different test functions. Clicking on the soft key SET in the REV Control Panel opens the Setup & Control window. The Setup & Control window allows performing more advanced operations on the receiver like Clear-Scan, Analyze and Sound Check operations.

Element	**Description**
Clear Scan	This tab provides option for performing Clear Scan All and Clear Scan group operations on the receiver
Clear Scan Band	This tab provides option for performing Clear Scan Band operations on the receiver T
Analyze	This tab provides option for performing Analyze operations on the receiver
Misc	This tab provides option for performing Sound check and other miscellaneous operations on the receiver

ClearScan

ClearScan All or ClearScan group operation can be performed from this tab. The operations can be performed only if the application is online. This tab is displayed by default when the dialog is opened. Options selected as Scan All pressing start button starts the ClearScan All operation and a progress bar gets displayed. The progress bar disappears and the results get displayed once the scan all operation is complete on the receiver. Click on stop when a ClearScan All/ Group operation is in progress aborts the operation.

Results of ClearScan All operation

Clicking on the one of the rows displayed automatically changes the scan option to scan group. Pressing start perform a scan group operation on the currently selected group.

ClearScan group operation can also be performed as an isolated operation and not as a continuation to the ClearScan All operation. In this case the scan operation is performed on the currently selected group.

Results of ClearScan Group operation

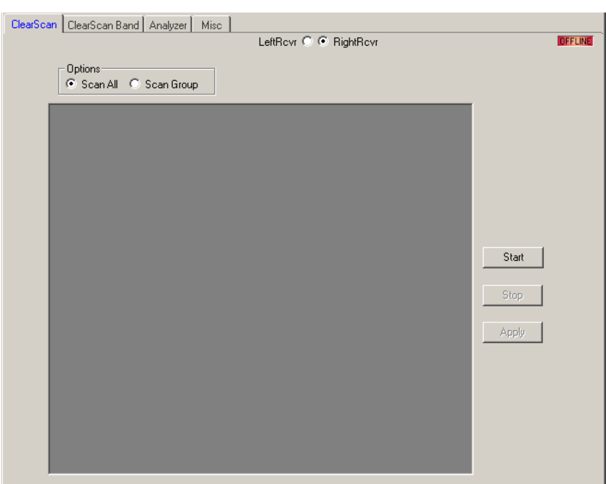
The result displayed is in the descending order of the clarity of the channels available in that group.

Click on one of the cells displayed. Press Apply to apply the currently selected group and channel to the receiver. Apply All option is enabled only if all the following conditions are satisfied:

- All the REV receivers present in the CAN bus have the same frequency band of operation
- The number of REV receivers present in the CAN bus is lesser than or equal to the number of free channels present in that group.

HINT: A REVD is counted as 2 receivers.

Click on Apply All to apply the group and channel to all the receivers present in the CAN bus. The clearest channel gets applied in the ascending order of their CAN addresses i.e. the receiver with the least CAN address gets assigned with the most clearest channel in the group.

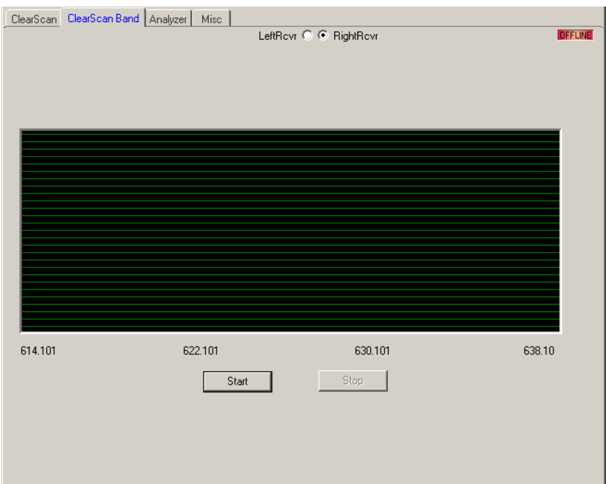


Element	Description
Online/Offline	The Online / Offline indicator signals whether the selected receiver is included in the network or off-line. The red OFFLINE indicator signals that the corresponding receiver is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding receiver is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
Left Rcvr / Right Rcvr	Select the left or the right receiver. **HINT: The receiver selection option is disabled in a REVS receiver.**
Scan All	Scan all o f the groups (factory and user)

Scan Group	Scan the group currently selected
Start	Start the scan of the selected receiver
Stop	Stop the scan of the selected receiver

ClearScan Band

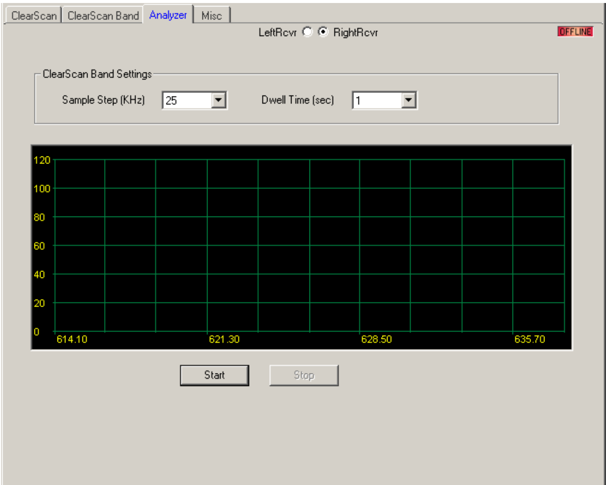
Clicking the ClearScan Band tab displays the option for performing the ClearScan Band operation on the receiver. Click Start to start the scan band operation. As this is a continuous operation the scan band operation will continue until it is manually stopped. Click Stop to stop the operation. This feature is useful for selecting one clear channel in a very busy RF environment.



Element	Description
Online/Offline	The Online / Offline indicator signals whether the selected receiver is included in the network or off-line. The red OFFLINE indicator signals that the corresponding receiver is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding receiver is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
Left Rcvr / Right Rcvr	Select the left or the right receiver. **HINT: The receiver selection option is disabled in a REVS receiver.**
Start	Start the scan of the selected receiver
Stop	Stop the scan of the selected receiver

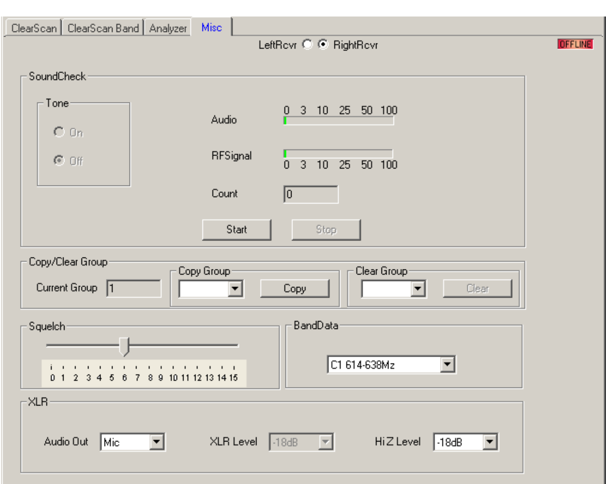
Analyzer

Clicking the Analyzer tab displays the option for performing the Analyze operation on the receiver. Click Start to start the Analyze operation. As this is a continuous operation it will continue until it is manually stopped. Click Stop to stop the operation. Sample Step and Dwell Time values can be modified from the list of values displayed in their respective combo-boxes.



Element	Description
Online/Offline	The Online / Offline indicator signals whether the selected receiver is included in the network or off-line. The red OFFLINE indicator signals that the corresponding receiver is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding receiver is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
Left Rcvr / Right Rcvr	Select the left or the right receiver. **HINT: The receiver selection option is disabled in a REVS receiver.**
Sample Step (kHz)	Set the sample step width in kHz.
Dwell Time (sec)	Set the dwell time in seconds.
Start	Start the scan of the selected receiver
Stop	Stop the scan of the selected receiver.

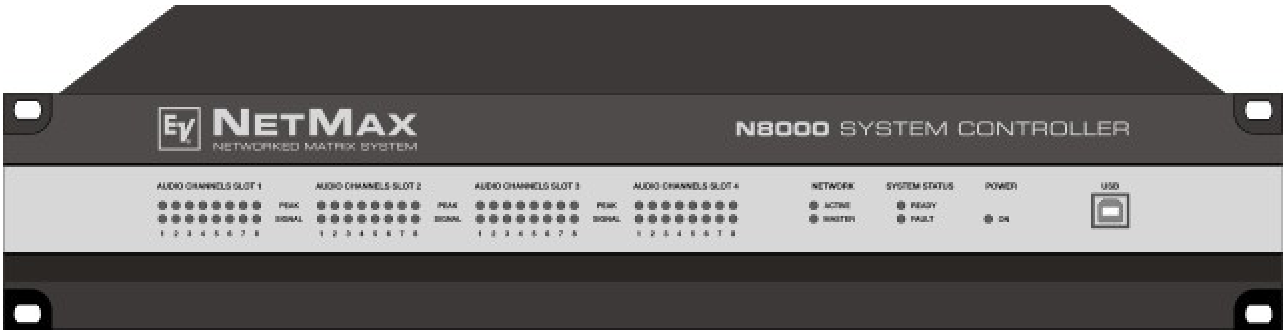
Misc



Element	Description
Online/Offline	The Online / Offline indicator signals whether the selected receiver is included in the network or off-line. The red OFFLINE indicator signals that the corresponding receiver is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding receiver is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately trans- mitted and active.
Left Rcvr / Right Rcvr	Select the left or the right receiver. **HINT: The receiver selection option is disabled in a REVS receiver.**
Tone	Select „On“ to activate the dropout tone. When the receiver increments the dropout counter, it will also send a 1 kHz audio tone out the out- put so you can hear where on the stage the drop occurs. Select „Off“ to turn off the tone.
Audio	This Peak Hold Audio Meter allows you to set the transmitter gain as high as possible for the application which maximizes the signal to noise ratio. Sing, yell or play the guitar at the loudest desired volume and adjust the gain so the meter peaks between 50 and 100.
RF Signal	This Low Hold RF Meter will tell you if you have adequate coverage in the performance area. If the RF level drops below 10 on the meter during a walk of the desired area, reposition the antennas, or change the channel and retest.
Count	Counter indicates the number of times the range was exceeded or some interference problems that was dealt with.
Start	Click Start to start the Sound check operation
Stop	Click Stop to stop the Sound check operation
Current Group	Current group indicates the group currently assigned to the receiver.
Copy Group	Copy group option lists the available user groups in the receiver.
Copy	Selecting a group and clicking this button copies a group in to the selected user group.
Clear Group	Clear group option is enabled only if the current group is a user group. Clear group option lists the available user groups in the receiver.
Clear	Selecting a group and clicking this button clears the selected user group.
Squelch	The squelch setting can be used to maximize range or immunity to noise.
Audio Out	Audio Out lists option for modifying the audio output to either „Mic“ or „Line“
XLR Level	Allows adjusting the signal level at the XLR output. **HINT: XLR Level is editable only if the Audio Out is selected as ‘Line’.**
Hi Z Level	Allows adjusting the signal level at the Hi Z output.

DIGITAL MATRIX

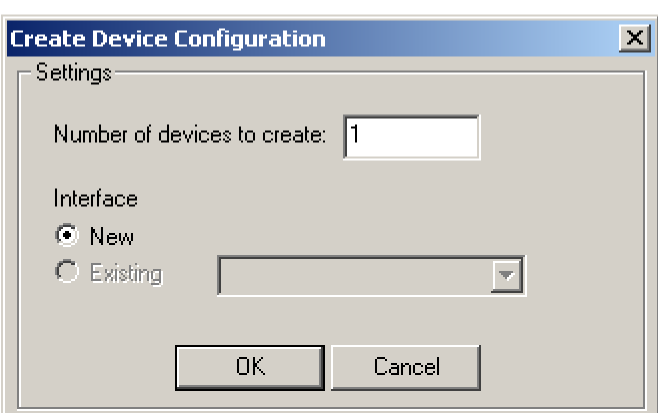
NetMax N8000 System Controller



NetMax is a modular, network-compatible and freely configurable audio system with which complete system solutions can be constructed. These system solutions exactly meet the customers' requirements. Applications are all kinds of professional audio installations, complex building sound reinforcement systems as well as concert sound applications. NetMax integrates all components ranging from the matrix to the speakers including system control and system monitoring in a common audio concept. The configuration, operation and monitoring of a NetMax system are handled by the IRIS-Net - Intelligent Remote & Integrated Supervision PC Software.
The central unit of NetMax is the N8000 system controller with up to 32 audio channels, mixer and matrix functions, signal processing and extensive control and monitoring functions. Several N8000 can be connected via a CobraNet or Dante audio network so that a large, decentralized audio system can be assembled.
NetMax also manages other Electro-Voice IRIS-Net-enabled devices, such as amplifiers, wireless microphones an external controllers. The connection is directly effected via CAN to N8000.
A NetMax system meets all relevant safety requirements. All audio connections, interfaces and processor systems are monitored and displayed in case of fault. By using CobraNet or Dante redundant networks can be assembled.

N8000 Device

Start by creating an N8000 Device in your IRIS-Net project. Drag an N8000 from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



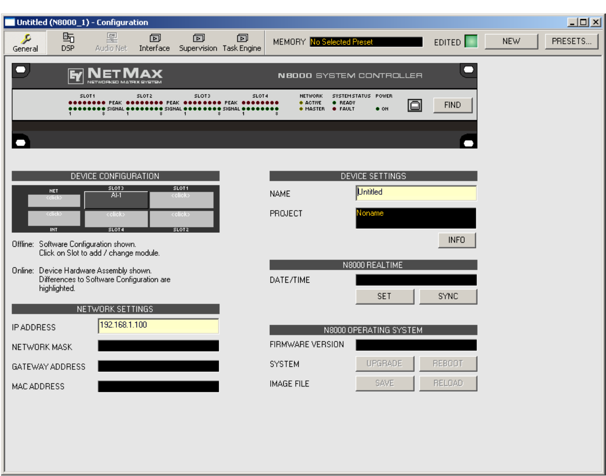
Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of N8000 Devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

Double clicking on an N8000 device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the N8000 device icon.
The following table lists all available N8000 dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	**Description**
General	This window allows hardware settings to be configured, e.g. input/output module slots, network settings, device name, system time and firmware version.
DSP	The DSP window lets you configure all DSP-parameters of the N8000.
Audio Net	This window provides detailed information and configuration of the CobraNet CM-1 module or Dante DM-1 module.
Interface	From this window the N8000 CAN bus, RS-232 ports and GPIO control port interfaces can all be configured. **HINT: Ethernet interface settings are explained under General dialog in the paragraph Network Settings.**
Supervision	This window provides an overview of the operational state and current fault status of the N8000.
Task Engine	This window lets you configure the N8000 Task Engine.

General Dialog

Double clicking on a N8000 by default opens the General dialog box. Here, the user can make basic settings that are necessary for flawless operation. All elements of the displayed N8000 front panel are active in on-line mode and correspond to the actual indicators on the unit.

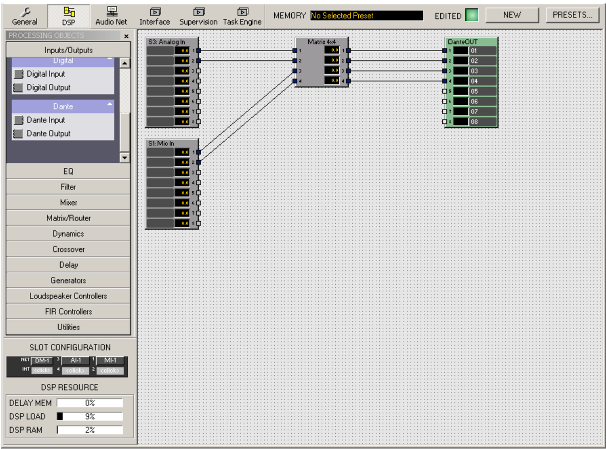


Element	**Description**
	A click on the FIND button lets the LEDs on the front panel of the N8000 blink. When on-line, this allows for easy identification of the N8000 the user communicates with at the moment.

	This view represents the rear panel of a N8000 with module slots and extension cards. In off-line mode, clicking onto the slots with the right mouse button and by exchanging, adding or deleting extension cards, allows defining the unit's configuration. When on-line, the display shows the actually installed extension cards. Differences from the off-line configuration are recognized and marked in yellow or red. **HINT: A yellow indicator signals that the hardware equipment differs from the software configuration. However, this difference does not cause any problems during on-line operation. A red indicator signals an existing conflict between hardware and software configurations, which needs to be remedied, either by customizing the N8000's hardware equipment or through modifying the software configuration.**
IP ADDRESS	Indicates the IP address of the N8000's Ethernet port (factory setting: 192.168.1.100). Enter the address of the N8000 with which you want to establish on-line communication.
NETWORK MASK	Indicates the Ethernet port's network mask (factory setting: 255.255.255.0).
GATEWAY ADDRESS	Indicates the standard gateway of the Ethernet port (factory setting: 192.168.1.1).
MAC ADDRESS	Indicates the MAC address of the connected N8000 when on-line. The MAC address of the N8000 is also shown on a label on the unit's rear panel.
NAME	IRIS-Net internal device name of the N8000.
PROJECT	IRIS-Net project filename.
INFO	This button displays information concerning the IRIS-Net project file.
DATE/TIME	Date and time of the N8000 system clock.
SET	This button opens the system clock settings dialog box.
SYNC	This button syncs the N8000 system clock to the PC system clock.
FIRMWARE VERSION	Indicates the firmware version of the N8000 when on-line.
REBOOT	Reboots the N8000.

DSP Dialog

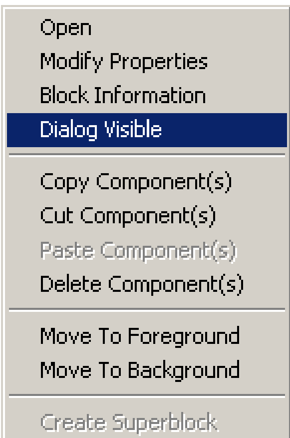
The DSP window lets the user configure all DSP functions of the NetMax N8000. This is possible by selecting DSP-Blocks of the Processing Objects categories on the left side of the screen and dragging them into the DSP worksheet. DSP- Blocks can be freely positioned and wired within the worksheet. Double clicking on a DSP-Block's icon allows editing its configuration and settings in detail.



Element	Description
MEMORY	The currently active Preset is shown here. Selecting a preset is possible from the Preset List in the Preset Manager.
EDITED	The EDITED indicator lights green if the momentarily active settings correspond to the Preset that had been loaded last. In case parameters of the loaded Preset have been changed, the EDITED indicator lights red.
STORE	Stores the DSP configuration’s current settings in the active Preset.
PRESETS...	Opens the Preset Manager.
SLOT CONFIGURATION NET <click> 3 <click> 1 AI-1 INT <click> 4 <click> 2 AO-1	Represents the hardware configuration of the N8000. Clicking with the right mouse button on one of the slots, when in off-line mode, allows editing the configuration. The indication represents the actual configuration when on-line. A red/yellow indicator signals differences between actual and off-line configuration (see also General Dialog).
DSP RESOURCE DELAY MEM 0% DSP LOAD 9% DSP RAM 2%	This indicates the DSP system’s estimated load. Adding supplementary DSP-Blocks is not possible when the actual load (DSP LOAD or DSP RAM) reaches 100%. Adding supplementary Delay-Blocks is not possible when the actual load (DELAY MEM) reaches 100%.

RIGHTS WHEN EDITING THE DSP CONFIGURATION

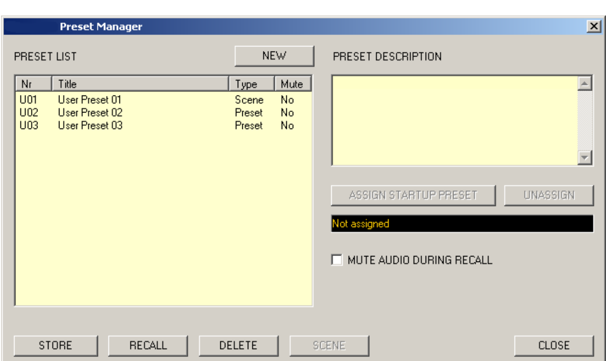
There are several ways to restrict the rights for editing the DSP configuration of a N8000. Generally, editing the DSP configuration is only possible when logged-in in IRIS-Net as administrator. Therefore, it is advisable to create additional user accounts in an IRIS-Net project, besides the administrator account, and to assign appropriate passwords. Please also refer to the corresponding chapter "Password Protection of a Project" on.
In addition, there is the option to determine for each single DSP block, whether a user that has no administrator rights may open the associated configuration dialog box.



To prevent access to the configuration dialog of a DSP block, you must be logged in as administrator. Open the contextual menu of the corresponding DSP block and deselect the contextual menu item "Dialog Visible".

PRESET MANAGER

The Preset Manager takes care of managing all N8000 presets. A preset holds all parameters of the current DSP configuration, e.g. equalizer settings, matrix nodes or delay values. Presets also hold the labels of input and output blocks, e.g. Analog Input, Analog Output, 8-Channel Mic Input, CobraNet Input and CobraNet Output. Labels of all other DSP-Blocks, e.g. matrix labels are not included.
Presets have no influence on the DSP configuration itself - amount, type and wiring of DSP-Blocks.

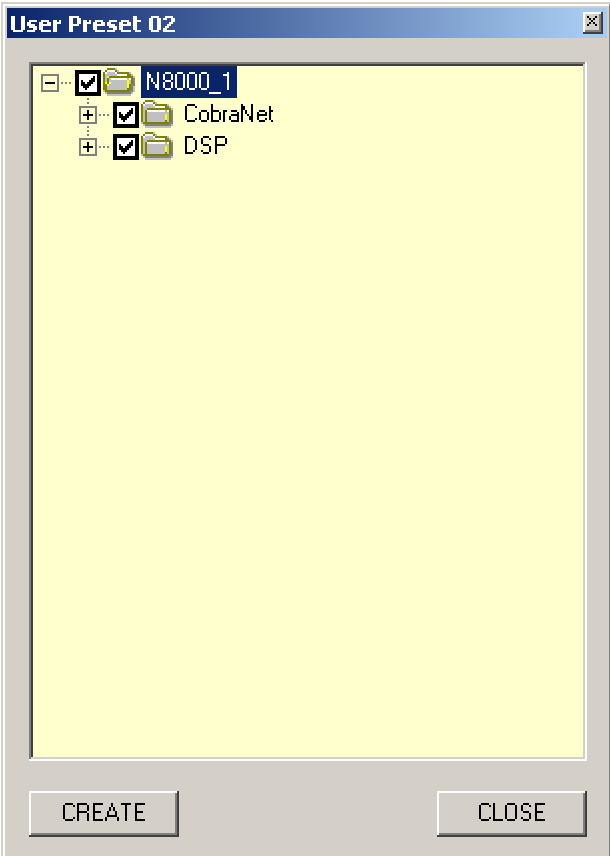


Element	Description
PRESET LIST	List of all NetMax N8000 presets. Selecting a preset with the left mouse button shows a description of the correspondent preset in the Preset Description window.

NEW	Adds a new preset to the preset list that holds the DSP configuration’s current settings. Up to 60 presets can be used.
Nr	Number of the preset. Up to 60 presets can be used.
Title	Name of the preset.
Type	A preset includes all dsp settings.
Mute	Audio outputs are muted during recall of the preset when the option MUTE AUDIO DURING RECALL is activated.
STORE	Stores the preset that you have selected and all current parameters in the preset list.
RECALL	Loads the selected preset into the preset list.
DELETE	Deletes the selected preset from the list.
PRESET DESCRIPTION	Shows a description of the selected preset.
ASSIGN STARTUP PRESET	The preset that you have selected in the preset list is automatically loaded upon power-on or restart of the N8000. With no startup-preset assigned, the N8000 starts using the settings that were active before it had been switched off. **HINT: If no start-up preset is assigned, under certain circumstances not all parameter changes can be restored after a restart of the N8000. In this case the audio output is muted after restart. Assigning a startup-preset is strongly recommended.**
UNASSIGN	Cancels the assignment of the previous start-preset.

SCENES

An existing preset can be converted into a scene. A scene contains a defined subset of the preset parameters. Select the preset to be converted from the Preset List and press the SCENE button. The following dialog box appears:



The dialog allows selecting which parameters that exist in the preset should be applied during loading. Parameters whose checkbox has not been selected are ignored.
When going online having the option Send All to Selected Devices selected the scenes are stored in N8000.

HINT: The Save Preset keyword of the N8000 or P 64 allows to save the edited parameters of an existing scene by pressing a Push Button (e.g. N8000_1.DSP.Savepreset=U01).

SUPERBLOCKS
Certain DSP blocks or Task Engine blocks that are frequently used in combination can be grouped together to form a Super Block, which, by definition, is available in the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016 and can be used like any normal DSP/Task Engine block.
The following information is stored in a Super Block:
- Number and type of an individual or several DSP/Task Engine blocks
- The wiring of the DSP/Task Engine blocks
- The parameters of the DSP/Task Engine blocks

Creating a new Super Block
Please proceed as follows to create a Super Block:
1. Create the desired DSP or Task Engine configuration exactly the same, as it is to be contained in the Super Block.
2. Mark all desired DSP/Task Engine blocks that are to be contained in the Super Block.
3. Right-click on one of the marked DSP/Task Engine blocks. The contextual menu appears.
4. Select "Create Super Block" from the contextual menu. The Super Block dialog appears.
5. Enter the desired name for the Super Block in the "Enter Super Block Name" dialog and click on OK.

The Super Block appears in the list of superblocks. It is stored in the IRIS-Net installation directory in the subdirectory "superblocks".

Use of a Super Block

Please proceed as follows to add a Super Block to the DSP configuration:

1. Open the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016. This category contains all available Super Blocks.
2. Drag the desired Super Block over to DSP/Task Engine configuration. The DSP/Task Engine configuration of the Super Block is displayed.
3. Move the added DSP/Task Engine blocks to the desired position. Now, the added DSP/Task Engine blocks can be used as usual.

Changing a Super Block

Please proceed as follows to change the DSP configuration of an existing Super Block:

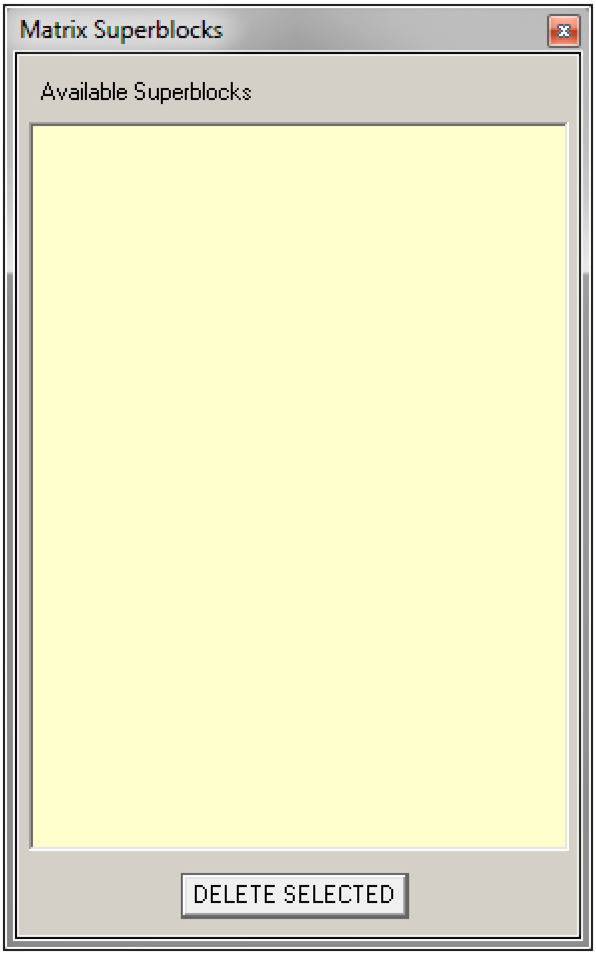
1. Open the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016. The category contains all available Super Blocks.
2. Drag the desired Super Block over to DSP/Task Engine configuration. The DSP/Task Engine configuration of the Super Block is displayed.
3. Apply the desired modifications to the DSP/Task Engine configuration.
4. Mark all DSP/Task Engine blocks that are to be contained in the changed Super Block.
5. Right-click on one of the marked DSP/Task Engine blocks. The contextual menu appears.
6. Select "Create Super Block" from the contextual menu. The "Super Block" dialog appears.
7. Enter the name of the existing Super Block in the "Enter Super Block Name" dialog and click on OK. An alert-message is displayed to inform the user that this Super Block already exists. Click on the dialog's YES button to confirm overwriting.

The Super Block now contains the changed DSP/Task Engine configuration.

Deleting a Super Blocks

Please proceed as follows to delete an existing Super Block:

Open the Matrix Superblocks dialog in the Menu Matrix, Superblocks (See Menu “Matrix” in *Menus, Commands and Symbol bar, page 77*).



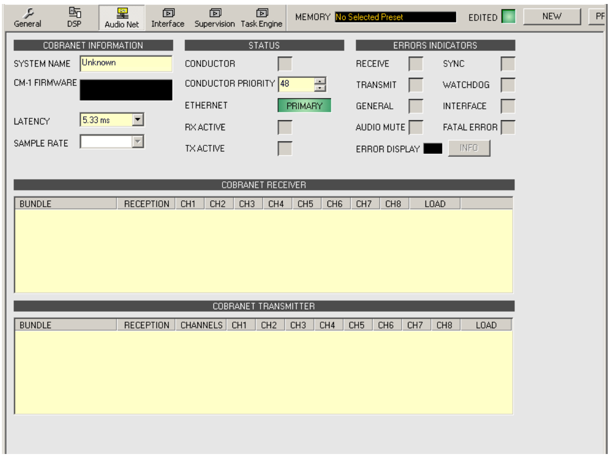
1. Select the Super Block to be deleted in the Available Superblocks list.
2. Click on the DELETE SELECTED button to delete the selected Super Block.

AudioNet Dialog

HINT: The AudioNet dialog is only accessible if a CM-1 or DM-1 is assembled in the General Dialog.

CM-1

This window provides detailed information about a CM-1 CobraNet module installed in the N8000. In addition, all received and transmitted bundles are listed in an overview. A CM-1 allows sending up to four bundles and receiving up to four bundles at the same time. The DSP blocks CobraNet Inputs provide the possibility to select a bundle to be received while the DSP blocks CobraNet Outputs allow configuring one bundle each for sending, including its contained channels.



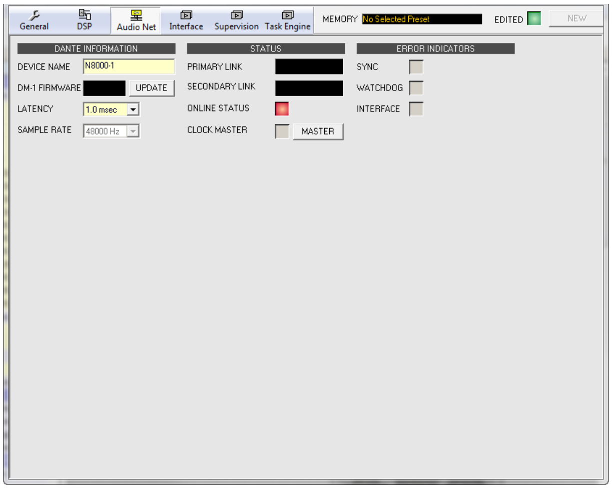
Element	Description
SYSTEM NAME	Alphanumeric name of the CM-1 module within CobraNet.
CM-1 FIRMWARE	Firmware version of the CM-1 module.
LATENCY	Latency setting for the CM-1 module. The settings 5.33 ms, 2.67 ms or 1.33 ms are available. The number of bundles that can be transmitted / received by the CM-1 depends on the selected latency: 5.33 ms: Up to 4 bundles (32 channels) can be transmitted and up to 4 bundles (32 channels) can be received simultaneously. 2.67 ms: Up to 4 bundles (32 channels) can be used (transmitted and/or received) in total. 1.33 ms: Up to 2 bundles (16 channels) can be used (transmitted and/or received) in total.
SAMPLE RATE	Sample rate of all CM-1 modules. Preset to 48 kHz.
CONDUCTOR	Green, if the CM-1 is the conductor (master) in the CobraNet, red, if another unit acts as conductor within the CobraNet.
CONDUCTOR PRIORITY	With a variety of units connected to a CobraNet, the unit that has the highest conductor priority automatically becomes the conductor. When setting the priority of a CM-1 to ”0” ensures that this CM-1 will never be the conductor in the network. Setting a CM-1’s priority to ”255” ensures that this CM-1 will always be the conductor in a network.
ETHERNET	Displays the Ethernet port (primary/secondary) of the CM-1 that is currently in use.
RX ACTIVE	Green, if data is being received via CobraNet; otherwise red.
TX ACTIVE	Green, if data is being transmitted via CobraNet; otherwise red.
RECEIVE	An error has been detected during data reception via CobraNet.
TRANSMIT	An error has been detected during data transmission via CobraNet.
GENERAL	System fault within the CM-1 module.

AUDIO MUTE	Audio transmission has been muted, because correct transmission cannot be guaranteed.
SYNC	Synchronizing the DSP system to CobraNet is not possible.
WATCHDOG	The CM-1 is being reset because of a hardware or software failure.
INTERFACE	An error has been detected in the communication with the CM-1 interface.
FATAL ERROR	A fatal error has been detected within the CM-1.
ERROR DISPLAY	Displays the corresponding error code for detected faults. 0 = no fault.
INFO	If an error code is being displayed, this button allows recalling information about the detected error.
COBRANET RECEIVER	
BUNDLE	Number of the received bundle.
RECEPTION	Indicates whether or not a bundle is currently received.
CHn	Displays the corresponding Resolutions of all channels of the received bundle.
LOAD	A bundle’s utilization ratio in percent. The load depends on the number and resolution of the channels transmitted in a bundle.
COBRANET TRANSMITTER	
BUNDLE	Number of the transmitted bundle.
RECEPTION	Indicates whether or not another unit receives the transmitted bundle.
CHANNELS	Number of channels transmitted in a bundle.
CHn	Displays the corresponding Resolutions of all channels of the transmitted bundle.
LOAD	A bundle’s utilization ratio in percent. The load depends on the number and resolution of the channels transmitted in a bundle.

DM-1

This window provides detailed information about a DM-1 Dante Interface module installed in the N8000. The DSP blocks Dante Inputs and Dante Outputs (see *DANTE INPUTS, page 383*) provide the possibility to select and configure Dante channels.

HINT: The Dante Configuration dialog (menu Tools > Dante Configuration) provides the possibility to configure the Dante network.

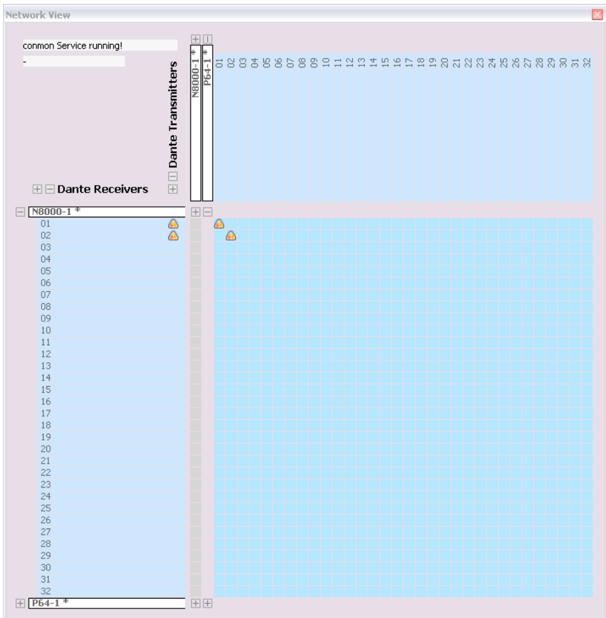


Element	**Description**
DEVICE NAME	Alphanumeric name of the N8000 within the Dante network.
DM-1 FIRMWARE	Firmware version of the DM-1 module.
UPDATE	The UPDATE button opens the web interface of the module.
LATENCY	Latency setting for the DM-1 module. The settings 0.5 ms, 1.0 ms or 5.0 ms are available.
SAMPLE RATE	Sample rate of all DM-1 modules. Preset to 48 kHz.
PRIMARY LINK	Indicates the Ethernet speed of the primary Ethernet interface.
SECONDARY LINK	Indicates the Ethernet speed of the secondary Ethernet interface.
ONLINE STATUS	Green, if the connection to the Dante network is OK; otherwise red;
CLOCK MASTER	Green, if the DM-1 is the clock master in the Dante Network, grey, if another unit acts as clock master within the Dante network.
MASTER	Press the MASTER button if this DM-1 should be clock master in the Dante network.
SYNC	Synchronizing the DSP system to Dante network is not possible.
WATCHDOG	The DM-1 is being reset because of a hardware or software failure.
INTERFACE	An error has been detected in the internal connection between N8000 and DM-1.

Network View

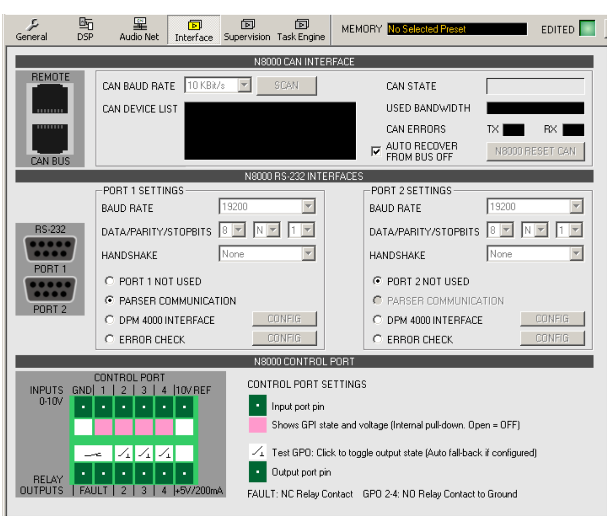
Click on Tools -> DanteConfiguration to open the Network Vies dialog. This dialog allows the configuration of transmitters and receivers in a Dante network. Left clicking the node in the matrix where the transmitter channel's column and the receiver channel's line meet with the mouse does connect an output to an input. Again clicking onto the corresponding node disconnects inputs and outputs.

Dante networks are subject to a restriction. Only one transmitter channel can be connected to a receiver channel at a time (mixing signals is not possible). However, connecting a transmitter channel to various receiver channels is possible.

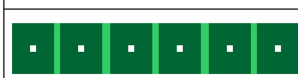


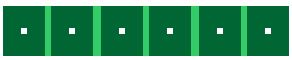
Interface Dialog

The Interface window allows configuring the different interfaces located on the rear panel of the N8000. All REMOTE CAN BUS, RS-232 and N8000 CONTROL PORT settings can be made in here. Configuring the Ethernet interface is done under Network Settings in the General window. Additionally, Ethernet settings are also accessible from the Matrix > Configuration via USB menu within IRIS-Net.



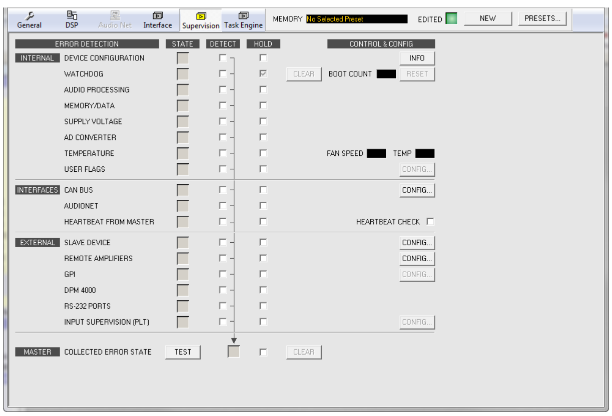
Element	Description
N8000 CAN INTERFACE	

CAN BAUD RATE	Transmission rate of the CAN-Bus. All devices on the CAN-Bus must be set to one common transmission rate. The SCAN button allows detecting the transmission rate of a CAN-Bus that is already in operation. Editing this parameter is possible in online mode only.
NUMBER OF DEVICES	The current number of devices on the CAN-Bus.
DEVICE ADDRESSES	Addresses of the devices that are currently connected to the CAN-Bus.
CAN DEVICE LIST	Opens the dialog box for configuring the connected devices.
CAN STATE	Displays the current CAN-Bus status. Possible indications are: BUS OK, Bus Heavy, Bus Off.
USED BANDWIDTH	Displays the used bandwidth of the CAN-Bus.
CAN ERRORS	Number of errors on the CAN-Bus that have been detected during send (TX) or reception (RX).
☑ AUTO RECOVER FROM BUS OFF	Option to automatically recover data transfer on the CAN-Bus after a Bus Off Condition.
RESET N8000 CAN	Resetting and reestablishing the connection between N8000 and CAN-Bus.
N8000 RS-232 INTERFACES	
BAUD RATE	RS-232 transmission rate.
DATA/PARITTY/STOPBITS	Data transmission parameter settings for data bit, parity bit and stop bit.
HANDSHAKE	Handshake settings.
◉ PORT 1 NOT USED	The RS-232 port is deactivated.
◉ PARSER COMMUNICATION	Accessing the N8000's ASCII Control Protocol is possible via RS-232 interface.
◉ DPM 4000 INTERFACE	Configuring the RS-232 port to act as PROMATRIX/PROANNOUNCE DPM 4000 interface. The Config button opens a window for further configuration.
◉ ERROR CHECK	Monitoring an external device via RS-232. The Config button opens a window for further configuration.
N8000 CONTROL PORT	
	Clicking with the right mouse button on the corresponding symbol of a control input provides opens the configuration dialog of this control input. (not yet activated)
	Displays the control inputs' current condition.

	It is possible to manually change the condition of the control outputs (Opener or Shutter). Depending on their according configuration, control outputs are only switched as long as the mouse button is pressed.
	Clicking with the right mouse button on the corresponding symbol of a control output provides opens the configuration dialog of this control output. (not yet activated)

Supervision Dialog

The Supervision window shows the condition of the NetMax N8000. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or included in the fault protocol. Further details can be set and are being displayed in the corresponding Config and Info dialogs.

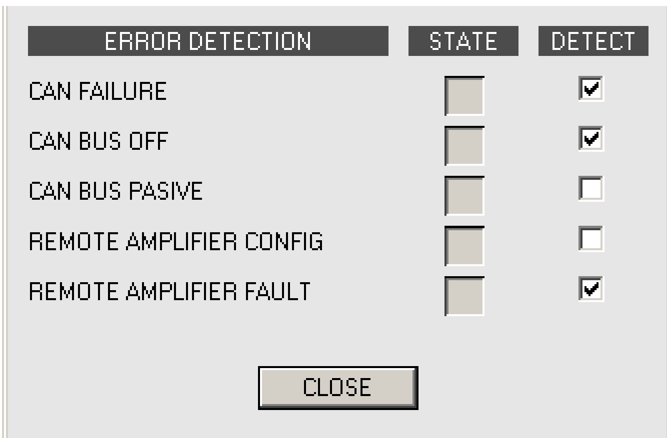


Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. The FAULT-LED on the front panel of the NetMax N8000 lights and the FAULT relay opens.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the feature is reset using the CLEAR button.
INTERNAL	
DEVICE CONFIGURATION	Error in the hardware configuration of the N8000. Pressing the INFO button reveals more detailed information concerning the error.
WATCHDOG	The Watchdog of the N8000 was activated. Press the CLEAR button to clear this error indication.
BOOT COUNT	Indicates the number of resets caused by the watchdog. Press the RESET button to reset the number to 0.

AUDIO PROCESSING	Error during the processing of audio data.
MEMORY/DATA	Memory or Read/Write error.
SUPPLY VOLTAGE	Error in the internal power supply unit.
AD CONVERTER	Malfunction of the control inputs’ A/D converters.
TEMPERATURE	Temperature overload of the N8000.
FAN SPEED	Current running speed of the N8000’s fan. Possible fan speeds are Off, Slow, Med. and High, see table below.
TEMP	Current temperature on the inside of the enclosure.
USER FLAGS	One or more User Flags have been set. CONFIG button for configuring User Flags.
INTERFACES	
CAN BUS	Fault condition on the CAN-Bus. The CONFIG button opens the CAN Interface Faults dialog, see below.
COBRANET	Fault condition on the CobraNet. Further details are provided via the CobraNet dialog box.
HEARTBEAT FROM MASTER	Query from the master N8000, which has been programmed to monitor this N8000, are not received anymore.
HEARTBEAT CHECK	Select this checkbox to check for heartbeat messages from other N8000s.
EXTERNAL	
SLAVE DEVICE	At least one NetMax N8000 that had to be monitored does not react anymore. The CONFIG button opens a list of N8000 that have been configured as Slave devices.
REMOTE AMPLIFIERS	A connected Remote Amplifier has transferred an error message. The CONFIG button opens the CAN Interface Faults dialog, see below.
GPI	The input voltage at a control input (GPI) is too high/low.
DPM 4000	The DYNACORD DPM 4000 that is connected via RS-232 port cannot be reached anymore.
RS-232 PORTS	Malfunction has been detected for an external device, which is being monitored via RS-232 port.
INPUT SUPERVISION (PLT)	Pilot tone recognition fault at the inputs of the N8000. Each input can separately be configured in the input blocks.
MASTER	
COLLECTED ERROR STATE	The FAULT-LED on the front panel of the N8000 lights at the occurrence of this type of error.

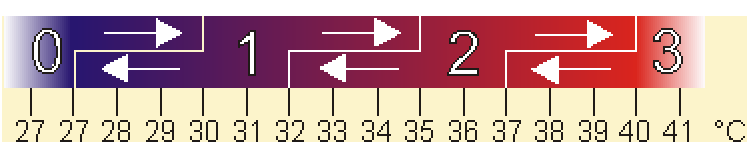
	Manually setting or resetting an error.
	Clears the indication of errors for which HOLD had been activated. Indication of still existing errors is not reset.

CAN INTERFACE FAULTS



Error	Description
CAN FAILURE	CAN self-testing was not successful. The CAN-Bus does not function.
CAN BUS OFF	The CAN-Bus is in the "Bus OFF" state.
CAN BUS PASSIVE	The CAN-Bus is in "Passive" mode.
REMOTE AMPLIFIER CONFIG	The RCM configuration does not represent the RCMs that are actually connected.
REMOTE AMPLIFIER FAULT	Collected Error for at least one RCM has been set.

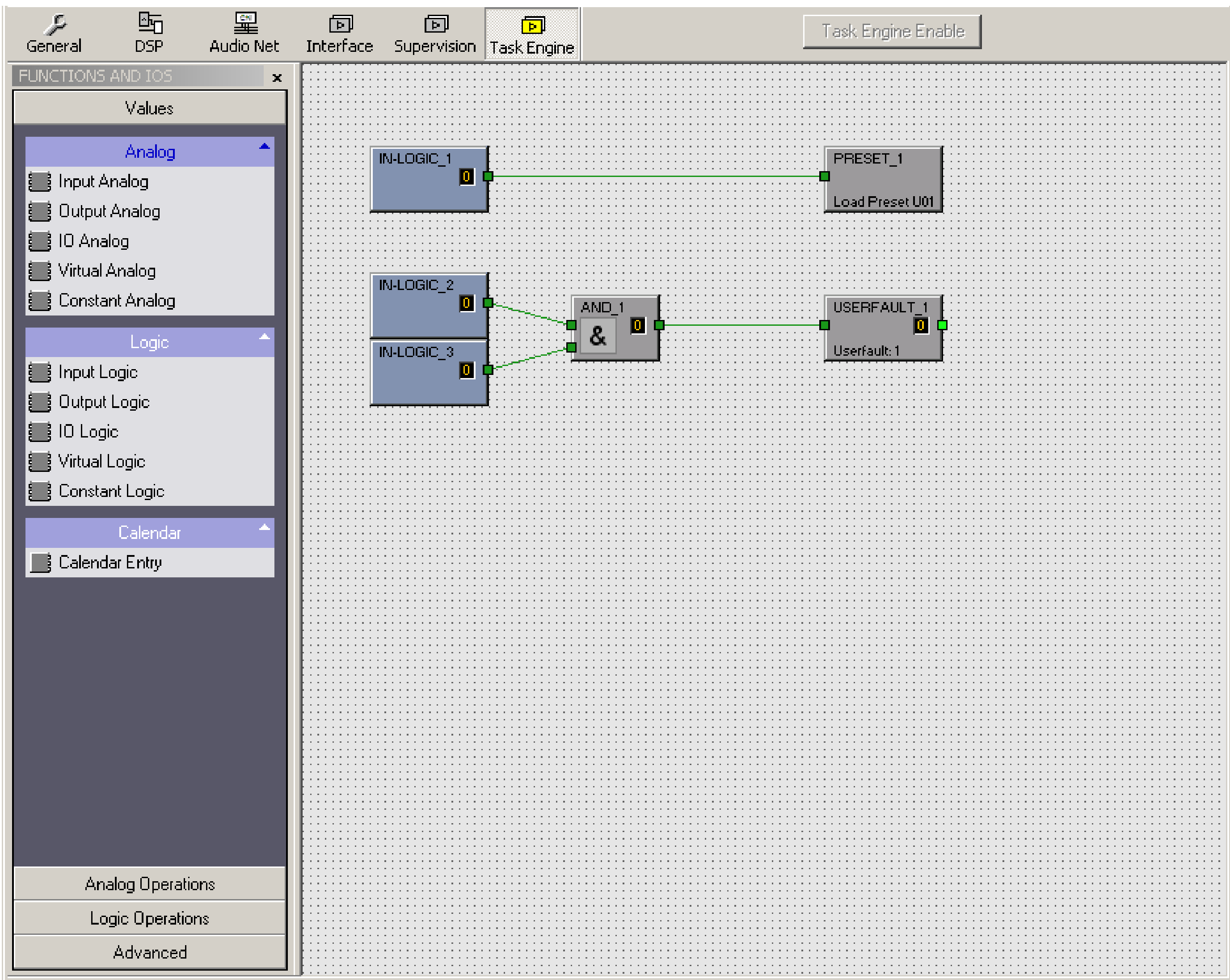
FAN SPEED



Speed	Description
0	Off
1	Slow
2	Med
3	High

Task Engine Dialog

The Task Engine Window allows configuring the Task Engine by dragging inputs, links or outputs from the categories of the FUNCTIONS AND IOS on the left corner of the screen into the Task Engine Worksheet. Elements can be freely positioned and wired within the worksheet. Double-clicking on inputs or outputs allows configuring them in detail. Building task engine configurations and changing the properties of task engine blocks is only possible in offline mode. If any changes are made the new configuration must be 'sent' to the N8000 when going online. Please refer to section “How to configure a Control” on page 20 how to assign functions and connections to a Task Engine block.

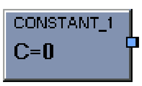
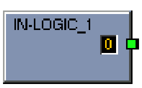
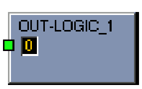
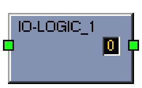
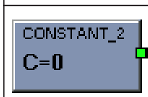


In the Task Engine, two classes of variables are available:

– Analog: variables of the type "analog" are rational numbers. Example: Level value (-80...+18) of a DSP block mono mixer output.
– Logic: variables of the type "logic" are Boolean values, i.e. only the values "0" and "1" are allowed. Example: Mute (0 = not muted, 1 = muted) of a DSP block mono mixer output.

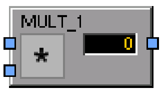
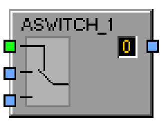
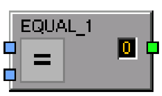
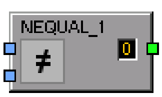
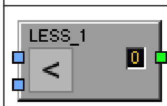
In the Task Engine, different colors are used to distinguish the two types of variables. Analog blocks have blue connection nodes and blue wiring connection lines. Logic blocks have green connection nodes and green wiring connection lines. Connecting analog nodes to logic nodes is not allowed.

VALUES

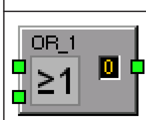
Element	Description
	The Input Analog block is a variable input parameter for rational numbers which always outputs the current value of the connection.
	The Output Analog block is an output parameter for rational numbers. The value at the input of this block is assigned to the connection.
	The IO Analog block is a variable input and output parameter for rational numbers. The value at the input of this block is assigned to the connection. The block always outputs the current value of the connection.
	The Virtual Analog block has the same behavior as the IO Analog block, but has no connection property. Instead the rational number assigned to the ‚Value' property of the block will be sent to the output.
	The Constant Analog block is a constant input parameter for a rational number. The rational number entered in the „Value" property of the block at Task Engine configuration time will always be sent to the output. This enables the block to provide a constant to other task engine blocks.
	The Input Logic block is a variable input parameter for Boolean values which always outputs the current value of the connection.
	The Output Logic block is an output parameter for Boolean values. The value at the input of this block is always assigned to the connection.
	The IO Logic block is a variable input and output parameter for Boolean values. The value at the input of this block is assigned to the connection. The block always outputs the current value of the connection.
	The Virtual Logic block has the same behavior as the IO Logic block, but has no connection property. Instead the logic value entered in the ‚Value' property of the block will be sent to the output.
	The Constant Logic block is a constant input parameter for a Boolean value. The Boolean value entered in the „Value" property of the block at Task Engine configuration time will always be sent to the output. This enables the block to provide a constant to other task engine blocks.
	The Calendar Entry blocks is a variable input parameter for Boolean values. The value at the output of this block depends on the configuration of the block and the system time of the N8000.

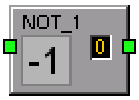
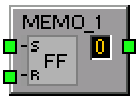
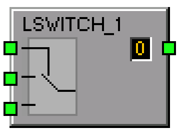
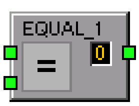
ANALOG OPERATIONS

Element	Description
	The rational number at the output of the Addition block is always the sum of rational numbers of the (wired) inputs. Not all inputs have to be wired.

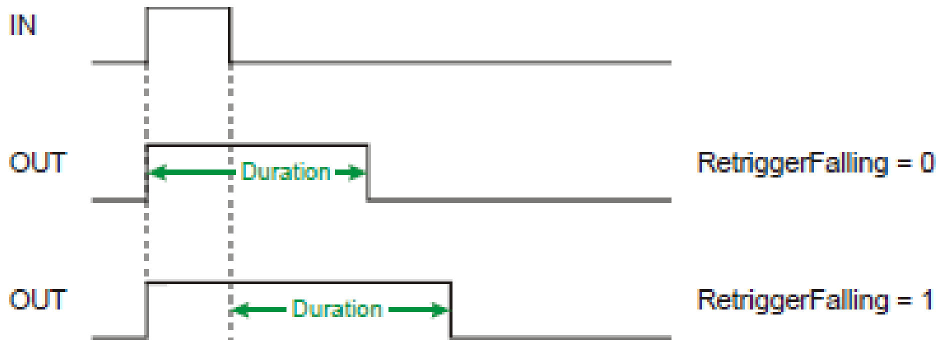
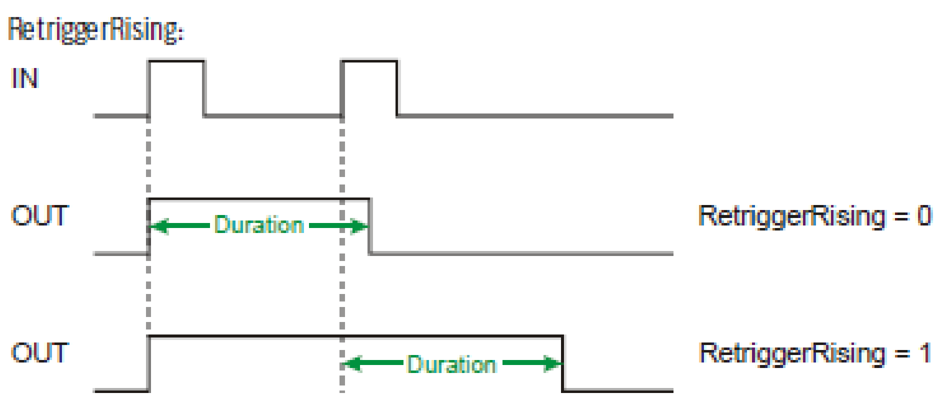
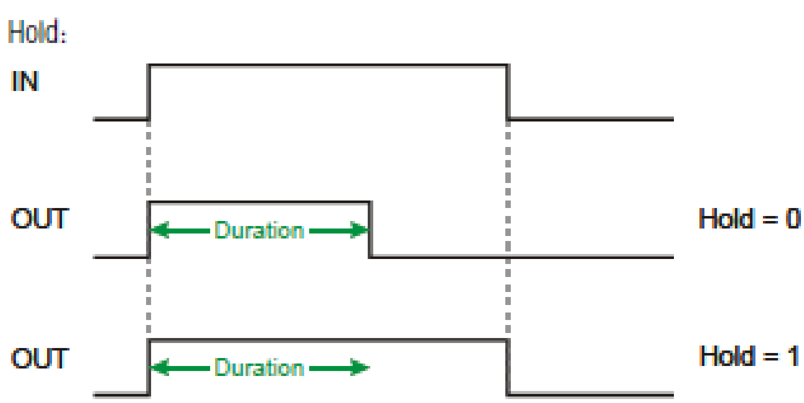
	The Subtraction block subtracts the rational number of the lower input from the rational number of the upper input. The output always presents the result of this arithmetic operation.
	The Multiplication block multiplies the rational number of the upper input with the rational number of the lower input. The output always presents the result of this arithmetic operation.
	The Division block divides the rational number of the upper input by the rational number of the lower input. **CAUTION: If the rational number "0" is present at the lower input, the rational number "0" is always output, independent of the upper input's value.**
	The Switch block switches the rational number at the center or lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when identical numbers are present at the inputs.
	The Not Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the numbers that are present at the inputs differ from each other.
	The Greater block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than the rational number of the lower input.
	The Greater Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than or equal to the rational number of the lower input.
	The Less block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is less than the rational number of the lower input.
	The Less Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is less than or equal to the rational number of the lower input.

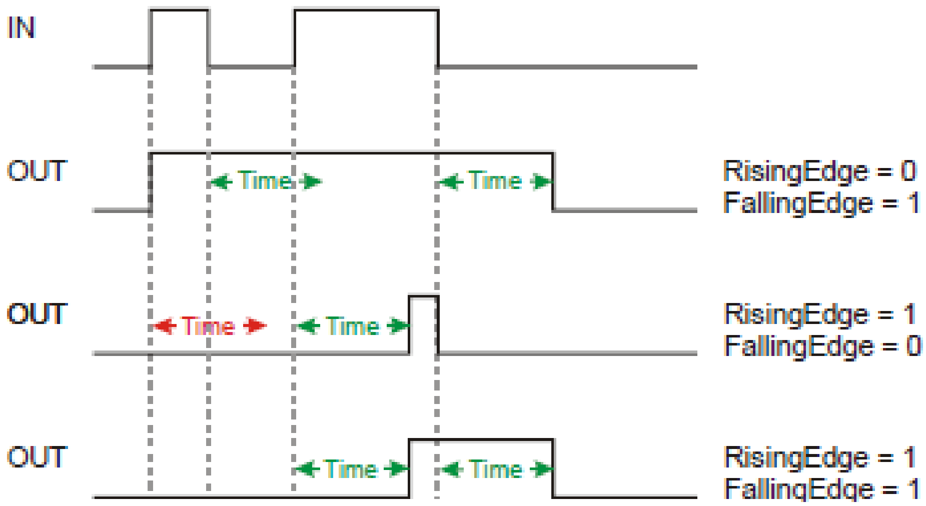
LOGIC OPERATIONS

Element	Description
	The Boolean value at the output of the AND block is only true when all (wired) inputs are true.
	The Boolean value at the output of the OR block is only true when at least one (wired) input is true.

	The Boolean value at the output of the XOR block is only true when the number of true (wired) inputs is odd.
	The NOT block negates the Boolean value at the input.
	The Memo (Flip-flop) block represents a Bistable flip-flop. The Boolean value at the output is set to true when the input S (Set) is true. The output stays true when the input S is false. To reset the output to false the input R (Reset) must be true.
	The Switch block switches the Boolean value at the center or the lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Load Preset block is used for loading the preset of a N8000 or P 64. The preset is loaded when the input is true.
	The User Fault block is used for indicating a Boolean value by a User Fault. The User Fault is active when the input is true. The output always presents the state of the User Fault.
	The Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are identical (i.e. both inputs are true or both inputs are false).
	The Not Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are different from each other (i.e. one input is true while the other input is false).

ADVANCED

Element	Description
 	The Timer block sets the Boolean value „State“ at the output to true for the time „Duration“, if the Boolean value at the input changes from false to true. Following charts illustrate the functionality of the available block parameters: **Retrigger Falling:**   

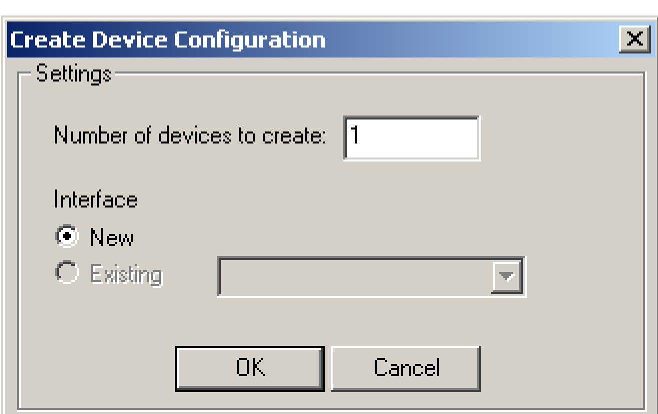
	The Debounce block is used for debouncing the Boolean value at the input. The debounce time is adjustable. Following chart illustrates the functionality of the available block parameters. One key function of the debounce block is that if the input changes from false to true and then back to false again before the debounce time has elapsed then the output will not be set to true. 
	The Analog Delay block is used for delaying rational numbers at the input. The value at the input will not be sent to the output until the delay time has elapsed. The maximum delay time is 20 seconds.
	The Logic Delay block is used for delaying Boolean values at the input. The value at the input will not be sent to the output until the delay time has elapsed. The maximum delay time is 20 seconds.
	The Text Box allows labeling the task engine configuration. Click the Modify Properties entry in the context menu to open the Edit Textbox dialog. This dialog allows editing the caption and e.g. font size and font type.
Superblocks	Superblocks are listed here. Please refer to page 240 how to use Superblocks.

P 64 Digital Matrix

The P 64 is a modular, network-compatible and freely configurable audio system with which complete system solutions can be constructed. These system solutions exactly meet the customers’ requirements. Applications are all kinds of professional audio installations, complex building sound reinforcement systems as well as concert sound applications.
The P 64 integrates all components ranging from the matrix to the speakers including system control and system monitoring in a common audio concept. The configuration, operation and monitoring of a P 64 system are effected by the IRIS-Net - Intelligent Remote & Integrated Supervision PC Software.
The P 64 includes up to 32 audio channels, mixer and matrix functions, signal processing and extensive control and monitoring functions. Several P 64 can be connected via a CobraNet or Dante audio network so that a large, decentralized audio system can be assembled.
The P 64 manages also the DYNACORD remote amplifiers including its speaker and system monitoring functions. The connection is directly effected via CAN to P 64.
The P 64 meets all relevant safety requirements. All audio connections, interfaces and processor systems are monitored and displayed in case of fault. By using CobraNet or Dante redundant networks can be assembled.

P 64 Device

Start by creating an P 64 Device in your IRIS-Net project. Drag an P 64 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

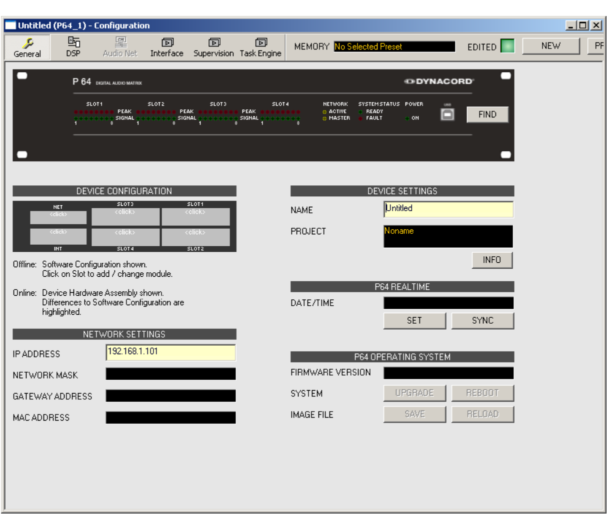


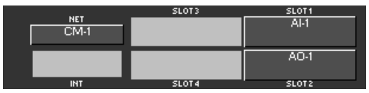
Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of P 64 devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an P 64 device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the P 64 device icon.
The following table lists all available P 64 dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	Description
General	This window allows hardware settings to be configured, e.g. input/output module slots, network settings, device name, system time and firmware version.
DSP	The DSP window lets you configure all DSP-parameters of the P 64.
Audio Net	This window provides detailed information and configuration of the CobraNet CM-1 module or Dante DM-1 module.
Interface	From this window the P 64 CAN bus, RS-232 ports and GPIO control port interfaces can all be configured. **HINT: Ethernet interface settings are explained under General dialog in the paragraph Network Settings.**
Supervisio n	This window provides an overview of the operational state and current fault status of the P 64.
Task Engine	This window lets you configure the P 64 Task Engine.

General Dialog

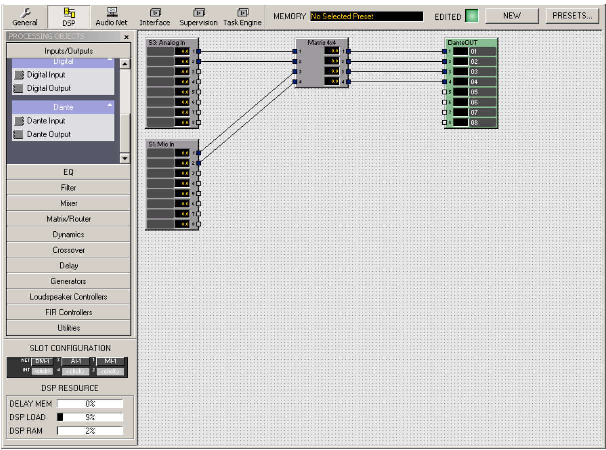
Double clicking on a P 64 by default opens the General dialog box. Here, the user can make basic settings that are necessary for flawless operation. All elements of the displayed P 64 front panel are active in on-line mode and correspond to the actual indicators on the unit.



Element	Description
FIND	A click on the FIND button lets the LEDs on the front panel of the P 64 blink. When on-line, this allows for easy identification of the P 64 the user communicates with at the moment.
	This view represents the rear panel of a P 64 with module slots and extension cards. In off-line mode, clicking onto the slots with the right mouse button and by exchanging, adding or deleting extension cards, allows defining the unit’s configuration. When on-line, the display shows the actually installed extension cards. Differences from the off-line configuration are recognized and marked in yellow or red. **HINT: A yellow indicator signals that the hardware equipment differs from the software configuration. However, this difference does not cause any problems during on-line operation. A red indicator signals an existing conflict between hardware and software configurations, which needs to be remedied, either by customizing the P 64’s hardware equipment or through modifying the software configuration.**
IP ADDRESS	Indicates the IP address of the P 64’s Ethernet port (factory setting: 192.168.1.100). Enter the address of the P 64 with which you want to establish on-line communication.
NETWORK MASK	Indicates the Ethernet port’s network mask (factory setting: 255.255.255.0).
GATEWAY ADDRESS	Indicates the standard gateway of the Ethernet port (factory setting: 192.168.1.1).
MAC ADDRESS	Indicates the MAC address of the connected P 64 when on-line. The MAC address of the P 64 is also shown on a label on the unit’s rear panel.
NAME	IRIS-Net internal device name of the P 64.
PROJECT	IRIS-Net project filename.
INFO	This button displays information concerning the IRIS-Net project file.
DATE/TIME	Date and time of the P 64 system clock.
SET	This button opens the system clock settings dialog box.
SYNC	This button syncs the P 64 system clock to the PC system clock.
FIRMWARE VERSION	Indicates the firmware version of the P 64 when on-line.
REBOOT	Reboots the P 64.

DSP Dialog

The DSP window lets the user configure all DSP functions of the P 64. This is possible by selecting DSP-Blocks of the Processing Objects categories on the left side of the screen and dragging them into the DSP worksheet. DSP-Blocks can be freely positioned and wired within the worksheet. Double clicking on a DSP-Block's icon allows editing its configuration and settings in detail.

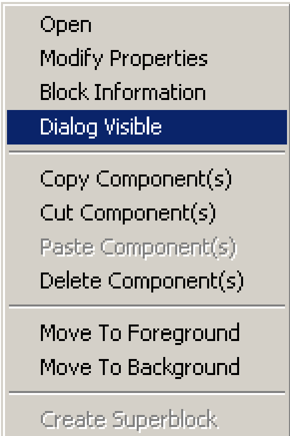


Element	Description
MEMORY	The currently active Preset is shown here. Selecting a preset is possible from the Preset List in the Preset Manager.
EDITED	The EDITED indicator lights green if the momentarily active settings correspond to the Preset that had been loaded last. In case parameters of the loaded Preset have been changed, the EDITED indicator lights red.
STORE	Stores the DSP configuration's current settings in the active Preset.
PRESETS...	Opens the Preset Manager.
SLOT CONFIGURATION NET <click> 3 <click> 1 AI-1 INT <click> 4 <click> 2 AO-1	Represents the hardware configuration of the P 64. Clicking with the right mouse button on one of the slots, when in off-line mode, allows editing the configuration. The indication represents the actual configuration when on-line. A red/yellow indicator signals differences between actual and off-line configuration (see also General Dialog).
DSP RESOURCE DELAY MEM 0% DSP LOAD 9% DSP RAM 2%	This indicates the DSP system's estimated load. Adding supplementary DSP-Blocks is not possible when the actual load (DSP LOAD or DSP RAM) reaches 100%. Adding supplementary Delay-Blocks is not possible when the actual load (DELAY MEM) reaches 100%.

RIGHTS WHEN EDITING THE DSP CONFIGURATION

There are several ways to restrict the rights for editing the DSP configuration of a P 64. Generally, editing the DSP configuration is only possible when logged-in in IRIS-Net as administrator. Therefore, it is advisable to create additional user accounts in an IRIS-Net project, besides the administrator account, and to assign appropriate passwords. Please also refer to the corresponding chapter "Password Protection Of A Project" on .

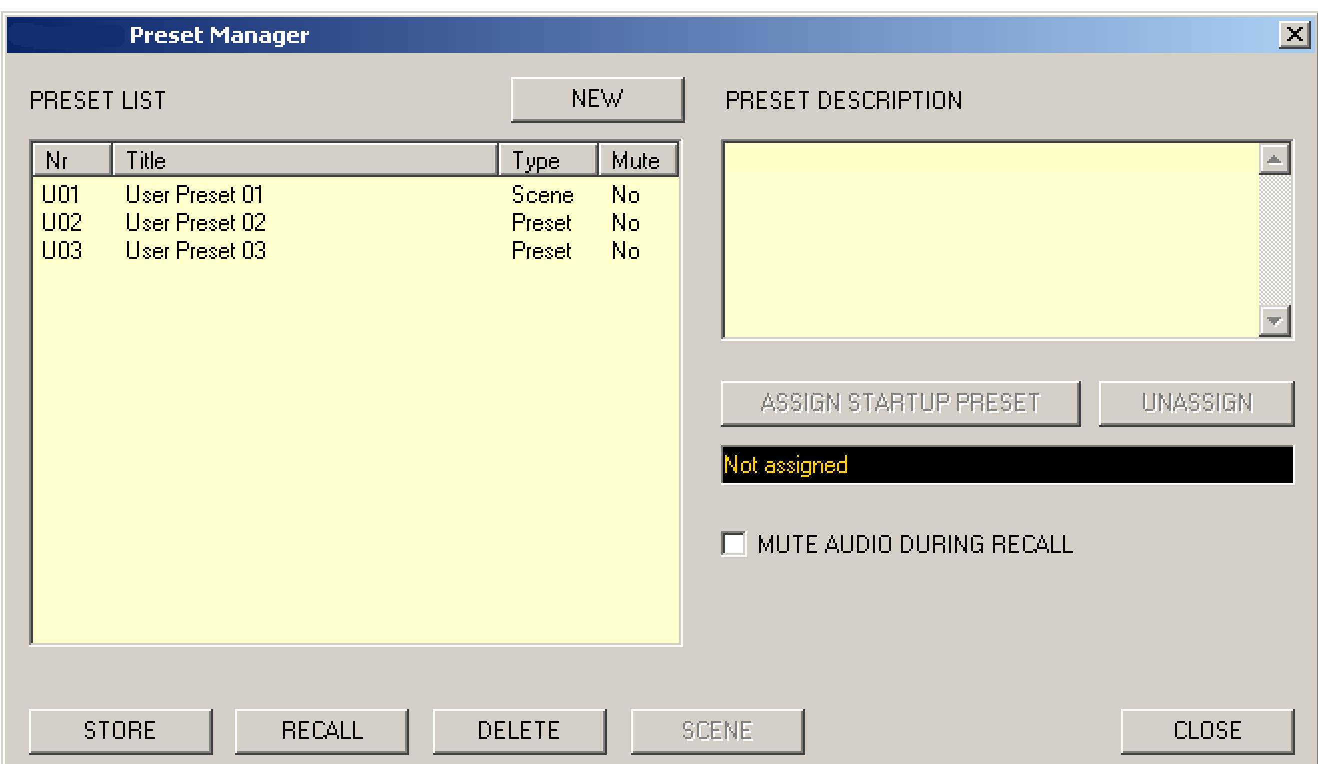
In addition, there is the option to determine for each single DSP block, whether a user that has no administrator rights may open the associated configuration dialog box.



To prevent access to the configuration dialog of a DSP block, you must be logged in as administrator. Open the contextual menu of the corresponding DSP block and deselect the contextual menu item "Dialog Visible".

PRESET MANAGER

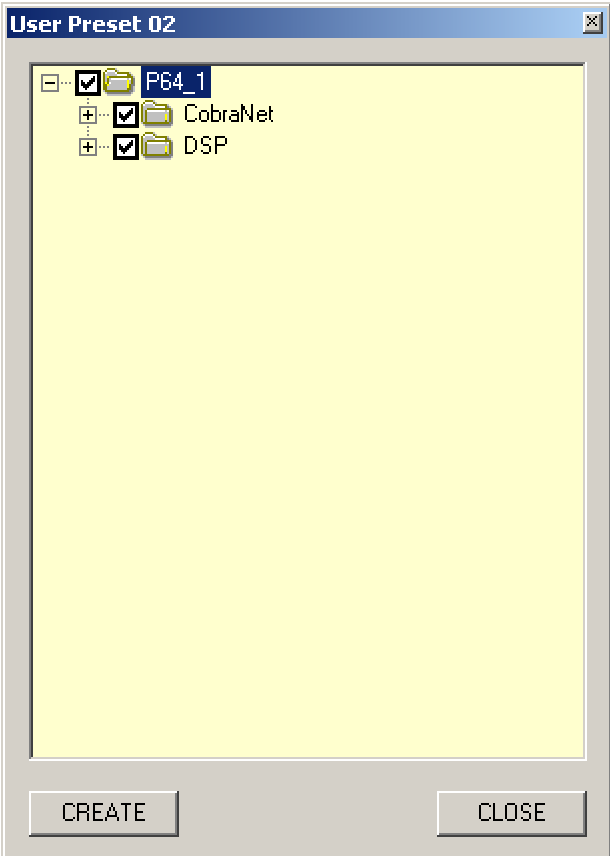
The Preset Manager takes care of managing all P 64 presets. A preset holds all parameters of the current DSP configuration, e.g. equalizer settings, matrix nodes or delay values. Presets also hold the labels of input and output blocks, e.g. Analog Input, Analog Output, 8-Channel Mic Input, CobraNet Input and CobraNet Output. Labels of all other DSP-Blocks, e.g. matrix labels are not included.
Presets have no influence on the DSP configuration itself - amount, type and wiring of DSP-Blocks.



Element	Description
PRESET LIST	List of all P 64 presets. Selecting a preset with the left mouse button shows a description of the correspondent preset in the Preset Description window.

NEW	Adds a new preset to the preset list that holds the DSP configuration’s current settings. Up to 60 presets can be used.
Nr	Number of the preset. Up to 60 presets can be used.
Title	Name of the preset.
Type	A preset includes all dsp settings.
Mute	Audio outputs are muted during recall of the preset when the option MUTE AUDIO DURING RECALL is activated.
STORE	Stores the preset that you have selected and all current parameters in the preset list.
RECALL	Loads the selected preset into the preset list.
DELETE	Deletes the selected preset from the list.
PRESET DESCRIPTION	Shows a description of the selected preset.
ASSIGN STARTUP PRESET	The preset that you have selected in the preset list is automatically loaded upon power-on or restart of the P 64. With no startup-pre- set assigned, the P 64 starts using the settings that were active before it had been switched off. **HINT: If no start-up preset is assigned, under certain circumstances not all parameter changes can be restored after a restart of the P 64. In this case the audio output is muted after restart. Assigning a startup-preset is strongly recommended.**
UNASSIGN	Cancels the assignment of the previous start-preset.

SCENES
An existing preset can be converted into a scene. A scene contains a defined subset of the preset parameters. Select the preset to be converted from the Preset List and press the SCENE button. The following dialog box appears:



The dialog allows selecting which parameters that exist in the preset should be applied during loading. Parameters whose checkbox has not been selected are ignored.
When going online having the option Send All to Selected Devices selected the scenes are stored in P 64.
HINT: The Save Preset keyword of the N8000 or P 64 allows to save the edited parameters of an existing scene by pressing a Push Button (e.g. N8000_1.DSP.Savepreset=U01).

SUPERBLOCKS
Certain DSP blocks or Task Engine blocks that are frequently used in combination can be grouped together to form a Super Block, which, by definition, is available in the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016 and can be used like any normal DSP/Task Engine block.
The following information is stored in a Super Block:
– Number and type of an individual or several DSP/Task Engine blocks
– The wiring of the DSP/Task Engine blocks
– The parameters of the DSP/Task Engine blocks

Creating a new Super Block
Please proceed as follows to create a Super Block:
1. Create the desired DSP or Task Engine configuration exactly the same, as it is to be contained in the Super Block.
2. Mark all desired DSP/Task Engine blocks that are to be contained in the Super Block.
3. Right-click on one of the marked DSP/Task Engine blocks. The contextual menu appears.
4. Select "Create Super Block" from the contextual menu. The Super Block dialog appears.
5. Enter the desired name for the Super Block in the "Enter Super Block Name" dialog and click on OK.

The Super Block appears in the list of superblocks. It is stored in the IRIS-Net installation directory in the subdirectory "superblocks".

Use of a Super Block

Please proceed as follows to add a Super Block to the DSP configuration:

1. Open the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016. This category contains all available Super Blocks.
2. Drag the desired Super Block over to DSP/Task Engine configuration. The DSP/Task Engine configuration of the Super Block is displayed.
3. Move the added DSP/Task Engine blocks to the desired position. Now, the added DSP/Task Engine blocks can be used as usual.

Changing a Super Block

Please proceed as follows to change the DSP configuration of an existing Super Block:

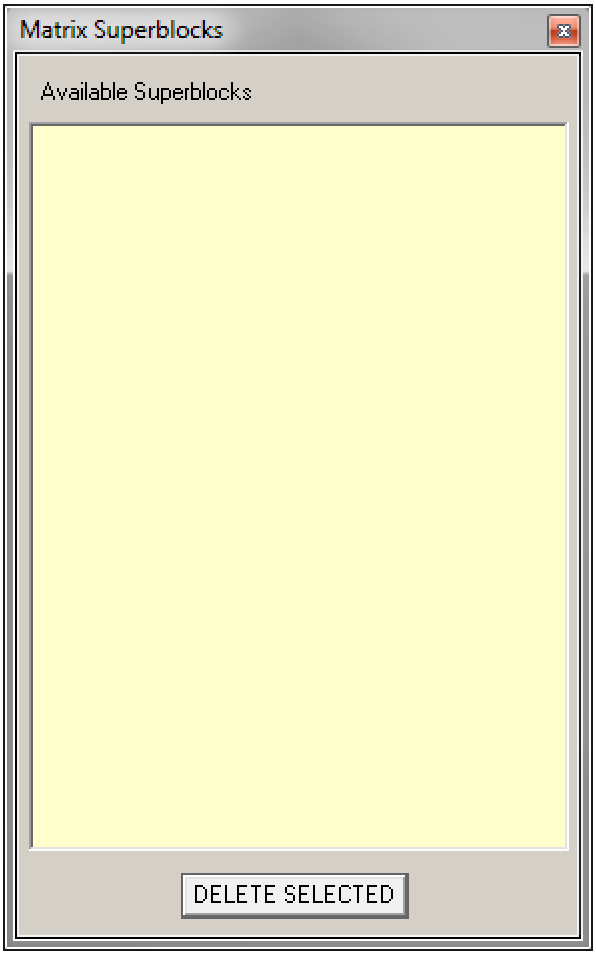
1. Open the category PROCESSING OBJECTS utilities of the N8000/P 64 or the category Advanced of the DPM 8016. The category contains all available Super Blocks.
2. Drag the desired Super Block over to DSP/Task Engine configuration. The DSP/Task Engine configuration of the Super Block is displayed.
3. Apply the desired modifications to the DSP/Task Engine configuration.
4. Mark all DSP/Task Engine blocks that are to be contained in the changed Super Block.
5. Right-click on one of the marked DSP/Task Engine blocks. The contextual menu appears.
6. Select "Create Super Block" from the contextual menu. The "Super Block" dialog appears.
7. Enter the name of the existing Super Block in the "Enter Super Block Name" dialog and click on OK. An alert-message is displayed to inform the user that this Super Block already exists. Click on the dialog's YES button to confirm overwriting.

The Super Block now contains the changed DSP/Task Engine configuration.

Deleting a Super Blocks

Please proceed as follows to delete an existing Super Block:

1. Open the Matrix Superblocks dialog in the Menu Matrix, Superblocks (see *Menus, Commands and Symbol bar, page 77*).



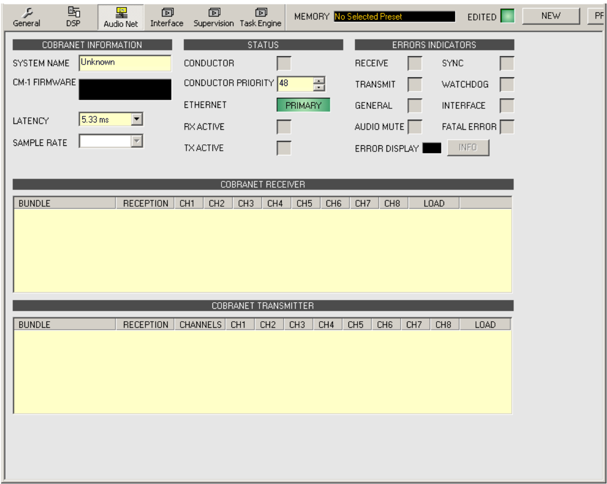
2. Select the Super Block to be deleted in the Available Superblocks list.
3. Click on the DELETE SELECTED button to delete the selected Super Block.

AudioNet Dialog

HINT: The AudioNet dialog is only accessible if a CM-1 or DM-1 is assembled in the General Dialog.

CM-1

This window provides detailed information about a CM-1 CobraNet module installed in the P 64. In addition, all received and transmitted bundles are listed in an overview. A CM-1 allows sending up to four bundles and receiving up to four bundles at the same time. The DSP blocks CobraNet Inputs provide the possibility to select a bundle to be received while the DSP blocks CobraNet Outputs allow configuring one bundle each for sending, including its contained channels.



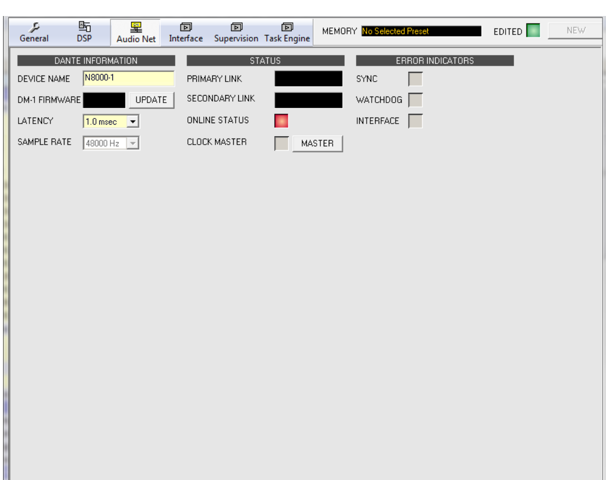
Element	**Description**
SYSTEM NAME	Alphanumeric name of the CM-1 module within CobraNet.
CM-1 FIRMWARE	Firmware version of the CM-1 module.
LATENCY	Latency setting for the CM-1 module. The settings 5.33 ms, 2.67 ms or 1.33 ms are available. The number of bundles that can be transmitted / received by the CM-1 depends on the selected latency: 5.33 ms: Up to 4 bundles (32 channels) can be transmitted and up to 4 bundles (32 channels) can be received simultaneously. 2.67 ms: Up to 4 bundles (32 channels) can be used (transmitted and/or received) in total. 1.33 ms: Up to 2 bundles (16 channels) can be used (transmitted and/or received) in total.
SAMPLE RATE	Sample rate of all CM-1 modules. Preset to 48 kHz.
CONDUCTOR	Green, if the CM-1 is the conductor (master) in the CobraNet, red, if another unit acts as conductor within the CobraNet.
CONDUCTOR PRIORITY	With a variety of units connected to a CobraNet, the unit that has the highest conductor priority automatically becomes the conductor. When setting the priority of a CM-1 to ”0” ensures that this CM-1 will never be the conductor in the network. Setting a CM-1’s priority to ”255” ensures that this CM-1 will always be the conductor in a network.
ETHERNET	Displays the Ethernet port (primary/secondary) of the CM-1 that is currently in use.
RX ACTIVE	Green, if data is being received via CobraNet; otherwise red.
TX ACTIVE	Green, if data is being transmitted via CobraNet; otherwise red.
RECEIVE	An error has been detected during data reception via CobraNet.
TRANSMIT	An error has been detected during data transmission via CobraNet.
GENERAL	System fault within the CM-1 module.
AUDIO MUTE	Audio transmission has been muted, because correct transmission cannot be guaranteed.
SYNC	Synchronizing the DSP system to CobraNet is not possible.
WATCHDOG	The CM-1 is being reset because of a hardware or software failure.
INTERFACE	An error has been detected in the communication with the CM-1 interface.

FATAL ERROR	A fatal error has been detected within the CM-1.
ERROR DISPLAY	Displays the corresponding error code for detected faults. 0 = no fault.
INFO	If an error code is being displayed, this button allows recalling information about the detected error.
COBRANET RECEIVER	
BUNDLE	Number of the received bundle.
RECEPTION	Indicates whether or not a bundle is currently received.
CHn	Displays the corresponding Resolutions of all channels of the received bundle.
LOAD	A bundle’s utilization ratio in percent. The load depends on the number and resolution of the channels transmitted in a bundle.
COBRANET TRANSMITTER	
BUNDLE	Number of the transmitted bundle.
RECEPTION	Indicates whether or not another unit receives the transmitted bundle.
CHANNELS	Number of channels transmitted in a bundle.
CHn	Displays the corresponding Resolutions of all channels of the transmitted bundle.
LOAD	A bundle’s utilization ratio in percent. The load depends on the number and resolution of the channels transmitted in a bundle.

DM-1

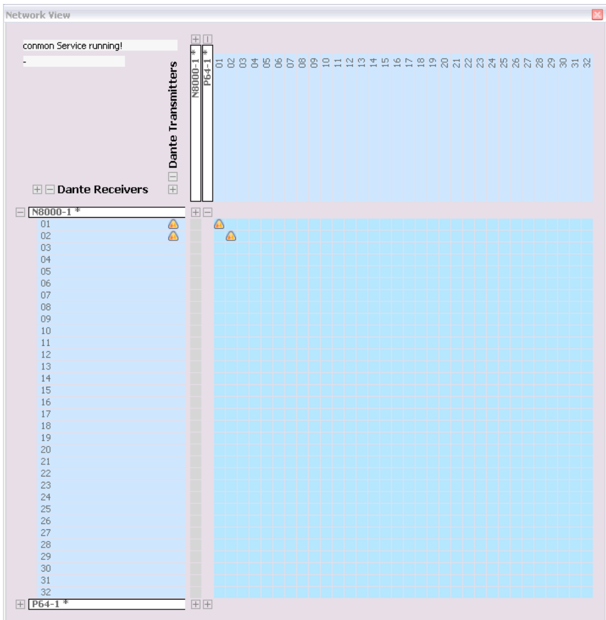
This window provides detailed information about a DM-1 Data Interface module installed in the P 64. The DSP blocks Dante Inputs and Dante Outputs (see *DANTE INPUTS, page 383*) provide the possibility to select and configure Dante channels.

HINT: The DanteConfiguration dialog (menu Tools > DanteConfiguration) provides the possibility to configure the Dante network.



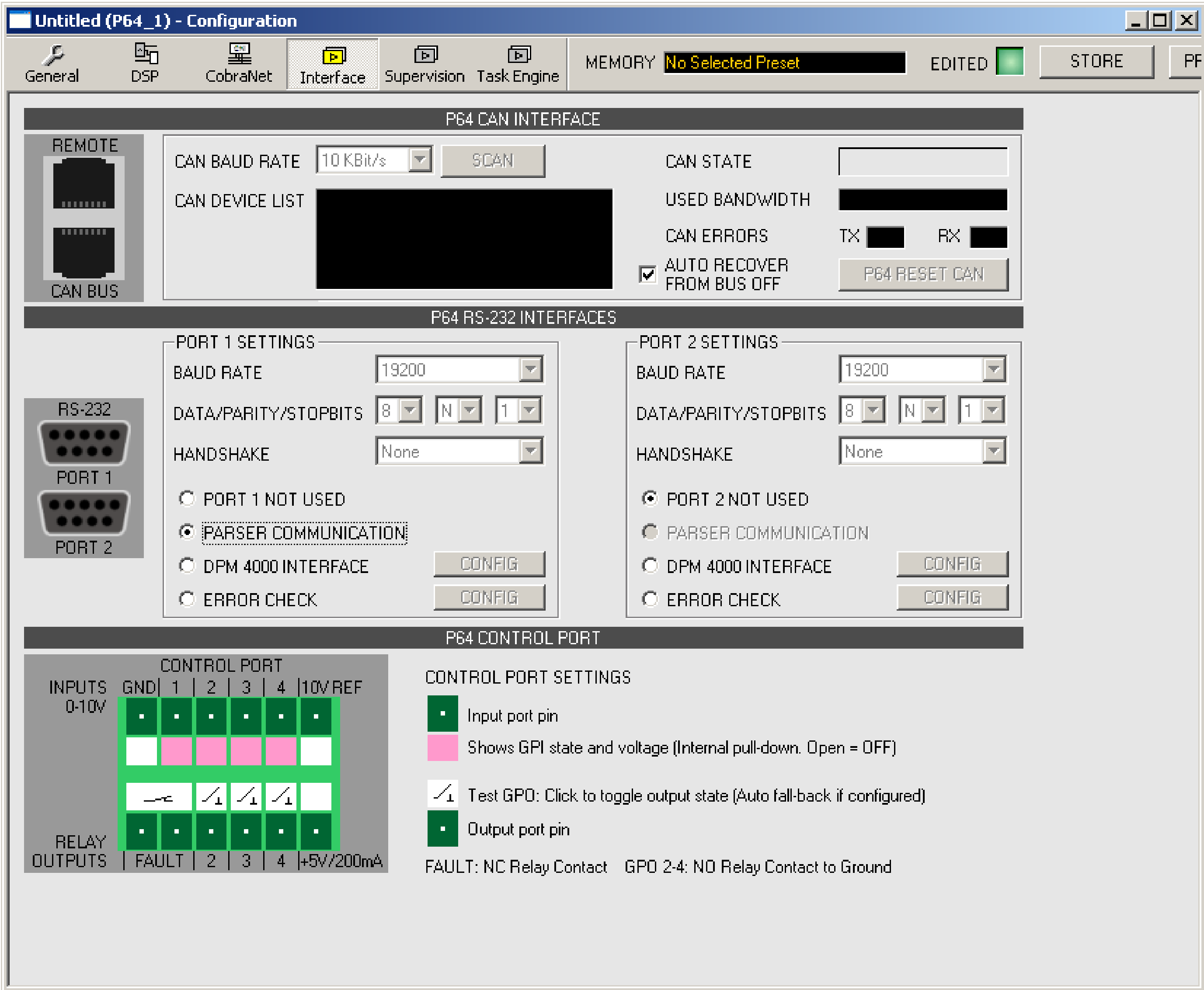
Element	Description
DEVICE NAME	Alphanumeric name of the P 64 within the Dante network.
DM-1 FIRMWARE	Firmware version of the DM-1 module.
	The UPDATE button opens the web interface of the module.
LATENCY	Latency setting for the DM-1 module. The settings 0.5 ms, 1.0 ms or 5.0 ms are available.
SAMPLE RATE	Sample rate of all DM-1 modules. Preset to 48 kHz.
PRIMARY LINK	Indicates the Ethernet speed of the primary Ethernet interface.
SECONDARY LINK	Indicates the Ethernet speed of the secondary Ethernet interface.
ONLINE STATUS	Green, if the connection to the Dante network is OK; otherwise red;
CLOCK MASTER	Green, if the DM-1 is the clock master in the Dante Network, grey, if another unit acts as clock master within the Dante network.
	Press the MASTER button if this DM-1 should be clock master in the Dante network.
SYNC	Synchronizing the DSP system to Dante network is not possible.
WATCHDOG	The DM-1 is being reset because of a hardware or software failure.
INTERFACE	An error has been detected in the internal connection between P 64 and DM-1.

Network View
Click on Tools -> DanteConfiguration to open the Network Vies dialog. This dialog allows the configuration of transmitters and receivers in a Dante network. Left clicking the node in the matrix where the transmitter channel's column and the receiver channel's line meet with the mouse does connect an output to an input. Again clicking onto the corresponding node disconnects inputs and outputs.
Dante networks are subject to a restriction. Only one transmitter channel can be connected to a receiver channel at a time (mixing signals is not possible). However, connecting a transmitter channel to various receiver channels is possible.

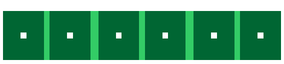
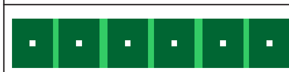


Interface Dialog

The Interface window allows configuring the different interfaces located on the rear panel of the P 64. All REMOTE CAN BUS, RS-232 and P 64 CONTROL PORT settings can be made in here. Configuring the Ethernet interface is done under Network Settings in the General window. Additionally, Ethernet settings are also accessible from the Matrix > Configuration via USB menu within IRIS-Net.

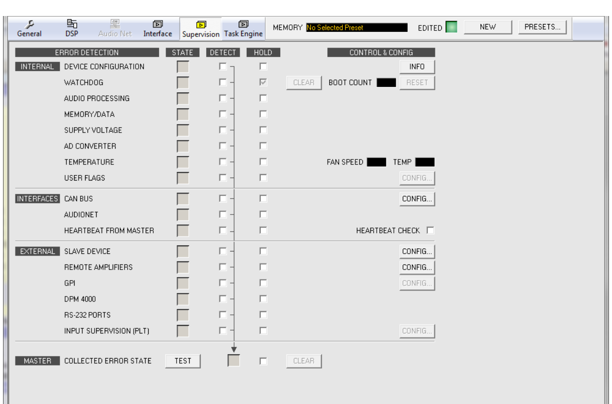


Element	Description
CAN BAUD RATE	Transmission rate of the CAN-Bus. All devices on the CAN-Bus must be set to one common transmission rate. The SCAN button allows detecting the transmission rate of a CAN-Bus that is already in operation. Editing this parameter is possible in online mode only.
NUMBER OF DEVICES	The current number of devices on the CAN-Bus.
DEVICE ADDRESSES	Addresses of the devices that are currently connected to the CAN-Bus.
DEVICE LIST	Opens the dialog box for configuring the connected devices.
CAN STATE	Displays the current CAN-Bus status. Possible indications are: BUS OK, Bus Heavy, and Bus Off.
USED BANDWIDTH	Displays the used bandwidth of the CAN-Bus.
CAN ERRORS	Number of errors on the CAN-Bus that have been detected during send (TX) or reception (RX).

AUTO RECOVER FROM BUS OFF	Option to automatically recover data transfer on the CAN-Bus after a Bus Off Condition.
RESET N8000 CAN	Resetting and reestablishing the connection between P 64 and CAN-Bus.
BAUD RATE	RS-232 transmission rate.
DATA/PARITTY/ STOPBITS	Data transmission parameter settings for data bit, parity bit and stop bit.
HANDSHAKE	Handshake settings.
PORT 1 NOT USED	The RS-232 port is deactivated.
PARSER COMMUNICATION	Accessing the P 64’s ASCII Control Protocol is possible via RS-232 interface.
DPM 4000 INTERFACE	Configuring the RS-232 port to act as PROMATRIX/PROANNOUNCE DPM 4000 interface. The Config button opens a window for further configuration.
ERROR CHECK	Monitoring an external device via RS-232. The Config button opens a window for further configuration.
	Clicking with the right mouse button on the corresponding symbol of a control input provides opens the configuration dialog of this control input. (not yet activated)
	Displays the control inputs’ current condition.
	It is possible to manually change the condition of the control outputs (Opener or Shutter). Depending on their according configuration, control outputs are only switched as long as the mouse button is pressed.
	Clicking with the right mouse button on the corresponding symbol of a control output provides opens the configuration dialog of this control output. (not yet activated)

Supervision Dialog

The Supervision window shows the condition of the P 64. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or included in the fault protocol. Further details can be set and are being displayed in the corresponding Config and Info dialogs.

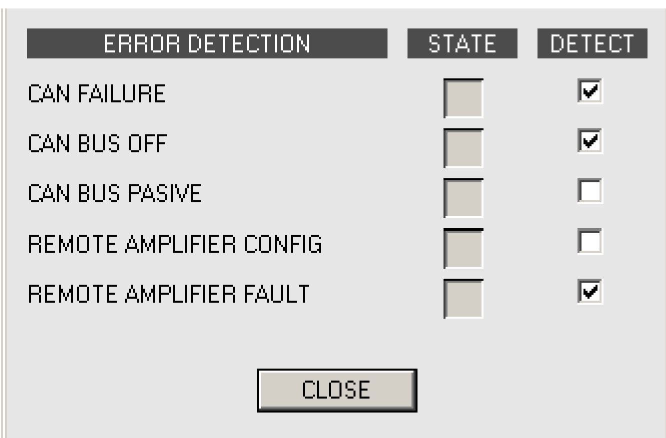


Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. The FAULT-LED on the front panel of the P 64 lights and the FAULT relay opens.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the feature is reset using the CLEAR button.
INTERNAL	
DEVICE CONFIGURATION	Error in the hardware configuration of the P 64. Pressing the INFO button reveals more detailed information concerning the error.
WATCHDOG	The Watchdog of the P 64 was activated. Press the CLEAR button to clear this error indication.
BOOT COUNT	Indicates the number of resets caused by the watchdog. Press the RESET button to reset the number to 0.
AUDIO PROCESSING	Error during the processing of audio data.
MEMORY/DATA	Memory or Read/Write error.
SUPPLY VOLTAGE	Error in the internal power supply unit.
AD CONVERTER	Malfunction of the control inputs’ A/D converters.
TEMPERATURE	Temperature overload of the P 64.
FAN SPEED	Current running speed of the P 64’s fan. Possible fan speeds are Off, Slow, Med. and High, see table below.
TEMP	Current temperature on the inside of the enclosure.
USER FLAGS	One or more User Flags have been set. CONFIG button for configuring User Flags.
INTERFACES	
CAN BUS	Fault condition on the CAN-Bus. The CONFIG button opens the CAN Interface Faults dialog, see below.
COBRANET	Fault condition on the CobraNet. Further details are provided via the CobraNet dialog box.
HEARTBEAT FROM MASTER	Query from the master P 64, which has been programmed to monitor this P 64, are not received anymore.
HEARTBEAT CHECK	Select this checkbox to check for heartbeat messages from other N8000s.
EXTERNAL	
SLAVE DEVICE	At least one P 64 that had to be monitored does not react anymore. The CONFIG button opens a list of P 64 that have been configured as Slave devices.
REMOTE AMPLIFIERS	A connected Remote Amplifier has transferred an error message. The CONFIG button opens the CAN Interface Faults dialog, see below.
GPI	The input voltage at a control input (GPI) is too high/low.
DPM 4000	The DYNACORD DPM 4000 that is connected via RS-232 port cannot be reached anymore.
RS-232 PORTS	Malfunction has been detected for an external device, which is being monitored via RS-232 port.

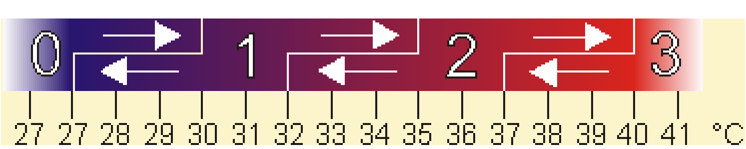
INPUT SUPERVISION (PLT)	Pilot tone recognition fault at the inputs of the P 64. Each input can separately be configured in the input blocks.
MASTER	
COLLECTED ERROR STATE	The FAULT-LED on the front panel of the P 64 lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.
CLEAR	Clears the indication of errors for which HOLD had been activated. Indication of still existing errors is not reset.

CAN INTERFACE FAULTS



Error	Description
CAN FAILURE	CAN self-testing was not successful. The CAN-Bus does not function.
CAN BUS OFF	The CAN-Bus is in the "Bus OFF" state.
CAN BUS PASSIVE	The CAN-Bus is in "Passive" mode.
REMOTE AMPLIFIER CONFIG	The RCM configuration does not represent the RCMs that are actually connected.
REMOTE AMPLIFIER FAULT	Collected Error for at least one RCM has been set.

FAN SPEED

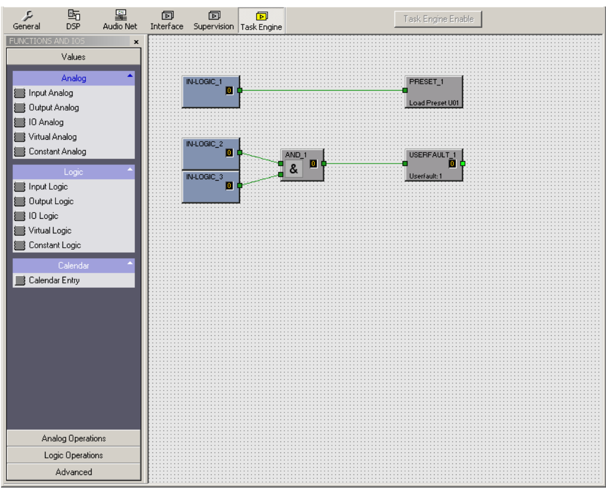


Speed	Description
0	Off
1	Slow

2	Med
3	High

Task Engine Dialog

The Task Engine Window allows configuring the Task Engine by dragging inputs, links or outputs from the categories of the FUNCTIONS AND IOS on the left corner of the screen into the Task Engine Worksheet. Elements can be freely positioned and wired within the worksheet. Double-clicking on inputs or outputs allows configuring them in detail. Building task engine configurations and changing the properties of task engine blocks is only possible in offline mode. If any changes are made the new configuration must be 'sent' to the P 64 when going online. Please refer to section “How to configure a Control” on page 20 how to assign functions and connections to a Task Engine block.

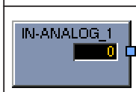
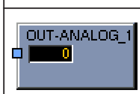
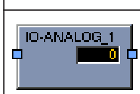


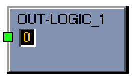
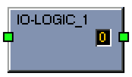
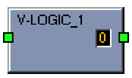
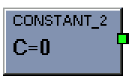
In the Task Engine, two classes of variables are available:

– Analog: variables of the type "analog" are rational numbers. Example: Level value (-80...+18) of a DSP block mono mixer output.
– Logic: variables of the type "logic" are Boolean values, i.e. only the values "0" and "1" are allowed. Example: Mute (0 = not muted, 1 = muted) of a DSP block mono mixer output.

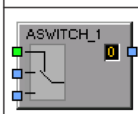
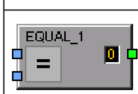
In the Task Engine, different colors are used to distinguish the two types of variables. Analog blocks have blue connection nodes and blue wiring connection lines. Logic blocks have green connection nodes and green wiring connection lines. Connecting analog nodes to logic nodes is not allowed.

VALUES

Element	**Description**
	The Input Analog block is a variable input parameter for rational numbers which always outputs the current value of the connection.
	The Output Analog block is an output parameter for rational numbers. The value at the input of this block is assigned to the connection.
	The IO Analog block is a variable input and output parameter for rational numbers. The value at the input of this block is assigned to the connection. The block always outputs the current value of the connection.
	The Virtual Analog block has the same behaviour as the IO Analog block, but has no connection property. Instead the rational number assigned to the ‚Value‘ property of the block will be sent to the output.

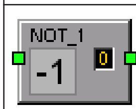
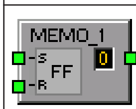
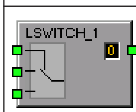
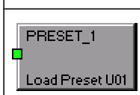
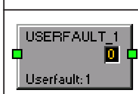
	The Constant Analog block is a constant input parameter for a rational number. The rational number entered in the „Value“ property of the block at Task Engine configuration time will always be sent to the output. This enables the block to provide a constant to other task engine blocks.
	The Input Logic block is a variable input parameter for Boolean values which always outputs the current value of the connection.
	The Output Logic block is an output parameter for Boolean values. The value at the input of this block is always assigned to the connection.
	The IO Logic block is a variable input and output parameter for Boolean values. The value at the input of this block is assigned to the connection. The block always outputs the current value of the connection.
	The Virtual Logic block has the same behaviour as the IO Logic block, but has no connection property. Instead the logic value entered in the ‚Value‘ pro- perty of the block will be sent to the output.
	The Constant Logic block is a constant input parameter for a Boolean value. The Boolean value entered in the „Value“ property of the block at Task Engine configuration time will always be sent to the output. This enables the block to provide a constant to other task engine blocks.
	The Calendar Entry blocks is a variable input parameter for Boolean values. The value at the output of this block depends on the configuration of the block and the system time of the P 64.

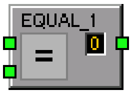
ANALOG OPERATIONS

Element	Description
	The rational number at the output of the Addition block is always the sum of rational numbers of the (wired) inputs. Not all inputs have to be wired.
	The Subtraction block subtracts the rational number of the lower input from the rational number of the upper input. The output always presents the result of this arithmetic operation.
	The Multiplication block multiplies the rational number of the upper input with the rational number of the lower input. The output always presents the result of this arithmetic operation.
	The Division block divides the rational number of the upper input by the rational number of the lower input. **CAUTION: If the rational number "0" is present at the lower input, the rational number "0" is always output, independent of the upper input's value.**
	The Switch block switches the rational number at the center or lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when identical numbers are present at the inputs.
	The Not Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the numbers that are present at the inputs differ from each other.

	The Greater block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than the rational number of the lower input.
	The Greater Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than or equal to the rational number of the lower input.
	The Less block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is less than the rational number of the lower input.
	The Less Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is less than or equal to the rational number of the lower input.

LOGIC OPERATIONS

Element	**Description**
	The Boolean value at the output of the AND block is only true when all (wired) inputs are true.
	The Boolean value at the output of the OR block is only true when at least one (wired) input is true.
	The Boolean value at the output of the XOR block is only true when the number of true (wired) inputs is odd.
	The NOT block negates the Boolean value at the input.
	The Memo (Flipflop) block represents a bistable flip-flop. The Boolean value at the output is set to true when the input S (Set) is true. The output stays true when the input S is false. To reset the output to false the input R (Reset) must be true.
	The Switch block switches the Boolean value at the center or the lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Load Preset block is used for loading the preset of a P 64. The preset is loaded when the input is true.
	The User Fault block is used for indicating a Boolean value by a User Fault. The User Fault is active when the input is true. The output always presents the state of the User Fault.

	The Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are identical (i.e. both inputs are true or both inputs are false).
	The Not Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are different from each other (i.e. one input is true while the other input is false).

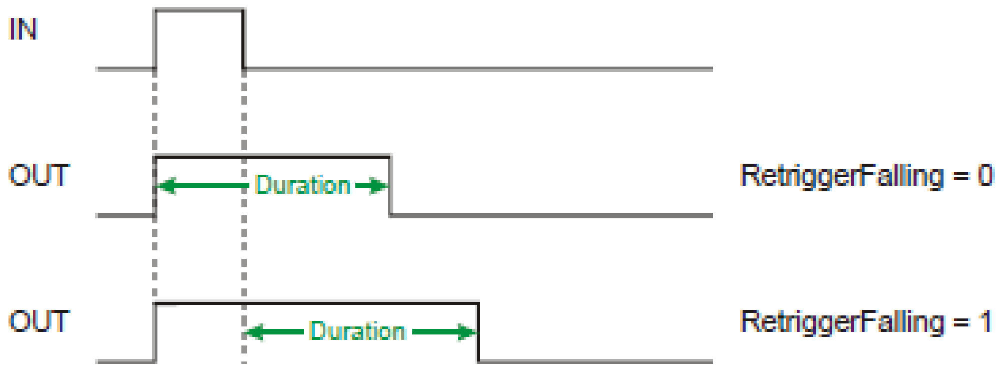
ADVANCED

<table>
<tr><th>Element</th><th>Description</th></tr>
<tr><td>

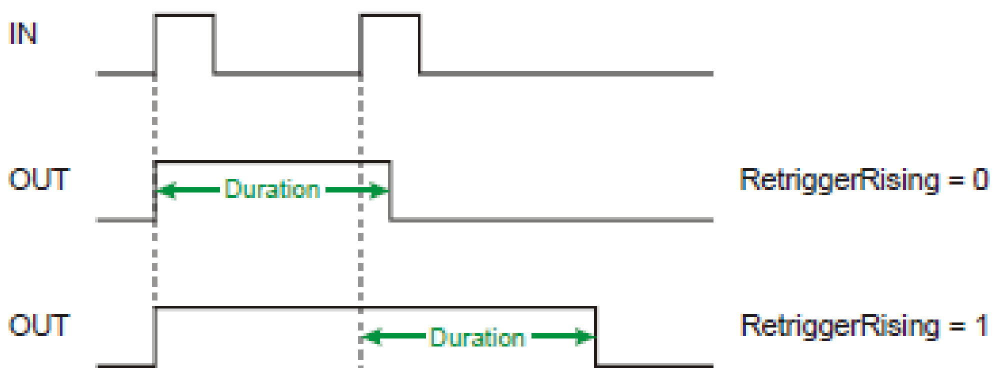
</td><td>

The Timer block sets the Boolean value „State“ at the output to true for the time „Duration“, if the Boolean value at the input changes from false to true. Following charts illustrate the functionality of the available block parameters:

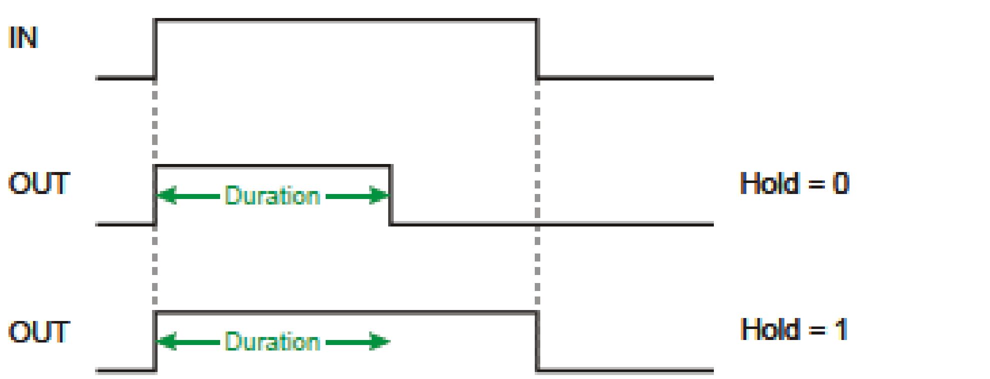
RetriggerFalling:



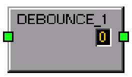
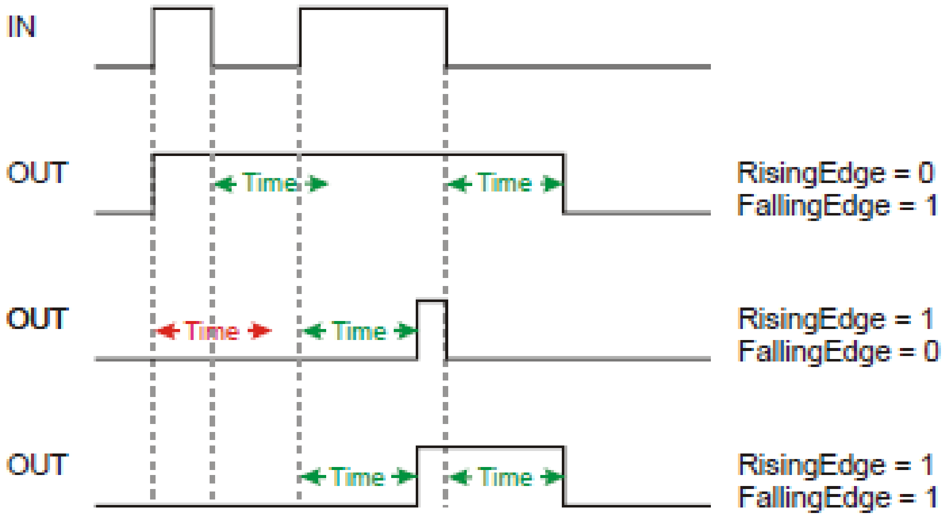
RetriggerRising:



Hold:



</td></tr>
</table>

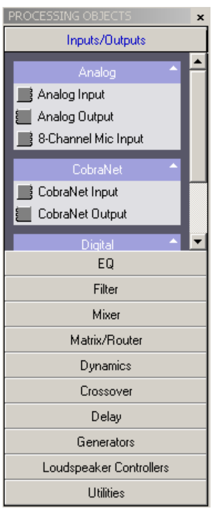
	The Debounce block is used for debouncing the Boolean value at the input. The debounce time is adjustable. Following chart illustrates the functionality of the available block parameters. One key function of the debounce block is that if the input changes from false to true and then back to false again before the debounce time has elapsed then the output will not be set to true. 
	The Analog Delay block is used for delaying rational numbers at the input. The value at the input will not be sent to the output until the delay time has elapsed. The maximum delay time is 20 seconds.
	The Logic Delay block is used for delaying Boolean values at the input. The value at the input will not be sent to the output until the delay time has elapsed. The maximum delay time is 20 seconds.
Text Box	The Text Box allows labeling the task engine configuration. Click the Modify Properties entry in the context menu to open the Edit Textbox dialog. This dialog allows editing the caption and e.g. font size and font type.
Superblocks	Superblocks are listed here. Please refer to page 240 how to use Superblocks.

N8000 and P 64

The following chapters are valid for N8000 and P 64.

DSP blocks

The library that holds all available DSP blocks (Processing Objects) is located on the left hand side of the DSP window. The DSP blocks are separated into groups or categories. Clicking on a group opens that group and provides access to the DSP blocks that are located inside. You can drag the desired DSP blocks from the list on to the worksheet, where they can be freely positioned and/or virtually wired with other blocks.



The available DSP blocks are summarized in following table.

Category	Description/ Definition	Inputs	Outputs	Description
Crossover	n-Way Crossover	1	1	Crossover blocks in Mono/Stereo with 1 to 5 channels. Each crossover channel has a High Pass and a Low Pass with the following parameters: type, frequency, polarity, level and mute.
			2	
			3	
			4	
			5	
	n-Way Stereo Crossover	2	2	
			4	
			6	
			8	
			10	
Delay	Mono Delay x ms	1	1	Delay blocks in Mono/Stereo with 10, 100, 500 or 2000 ms maximum delay. Entering values for distance and ambient temperature is possible.

	Stereo Delay x ms	2	2	
Dynamics	RMS-Compressor Mono	1	1	Compressor blocks in Mono/Stereo and RCM-24 structure. The compressors have adjustable thresholds, attack and release times, compression rates, some have Soft Knee characteristics, output level and side chain inputs.
	RMS-Compressor Stereo	2	2	
	RMS-Compressor (RCM-24)	1	1	
	Peak-Limiter Mono	1	1	Limiter blocks in Mono/Stereo, RCM-24 structure or Peak Anticipation algorithm. The limiters have adjustable thresholds, attack and release times.
	Peak-Limiter Stereo	2	2	
	Peak-Limiter (RCM-24)	1	1	
	PA-Limiter	1	1	
	AGC	1	1	Automatic Gain Control (AGC) in mono structure with adjustable threshold, compression rate, Soft Knee characteristic, hold and release times, target level and time constants for gain increase or decrease.
	ANC 2x1	1 PRM, 1 AMB	1	Ambient Noise Control (ANC) with adjustable threshold, hold and release times, multiple gain settings. Fader and Mute button for inputs.
	ANC 4x2	2 PRM, 2 AMB	2	
	ANC 8x4	4 PRM, 4 AMB	4	
	Ducker Mono	2	1	Ducker blocks in Mono/Stereo with adjustable thresholds, ducking level, attack, hold and release times. Fader and Mute button for inputs.
	Ducker Stereo	3	2	
	Expander	1	1	Expander block with adjustable threshold, attack and release times, ratio and output level.
	Gate	1	1	Gate block with adjustable threshold, attack, hold and release times and out- put level.

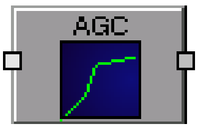
EQ - Graphic	n Band Graphic EQ	1	1	Graphic Equalizer in Mono/Stereo with 10, 15 or 31 bands. Two parametric EQs are provided as well.
	n Band Stereo Graphic EQ	2	2	
EQ - Parametric	n Band PEQ	1	1	Parametric Equalizer in Mono/Stereo with 1 to 32 bands. Freely selectable filter types (PEQ, Low/High-Shelving, Low/High/All Pass) with the following parameters: gain, frequency, quality/bandwidth or slope for each band.
	n Band Stereo PEQ	2	2	
Filter	FIR Filter	1	1	Finite Impulse Response blocks with order 256, 512, 768, 1024, 1280, 1536 or 1792. Possible filter types are Low Pass, High Pass or Band Pass.
	Low Pass Mono	1	1	Low Pass blocks in Mono/Stereo with the following parameters: frequency, slope and quality.
	Low Pass Stereo	2	2	
	High Pass Mono	1	1	High Pass blocks in Mono/Stereo with the following parameters: frequency, slope and quality.
	High Pass Stereo	2	2	
	Band Pass Mono	1	1	Band Pass blocks in Mono/Stereo with the following parameters: frequency and quality/bandwidth.
	Band Pass Stereo	2	2	
	Low Shelf Mono	1	1	Low Shelf blocks in Mono/Stereo with the following parameters: frequency, slope and gain.
	Low Shelf Stereo	2	2	
	High Shelf Mono	1	1	High Shelf blocks in Mono/Stereo with the following parameters: frequency, slope and gain.
	High Shelf Stereo	2	2	
	Notch Mono	1	1	Notch blocks in Mono/Stereo with the following parameters: frequency and quality/bandwidth.
	Notch Stereo	2	2	
	Allpass Mono	1	1	Allpass blocks in Mono/Stereo with the following parameters: frequency and order.
	Allpass Stereo	2	2	
	Tone Control Mono	1	1	Tone Control blocks in Mono/Stereo with gain controls for Bass/Mid/Treble.
	Tone Control Stereo	2	2	
FIR Controllers	n-Way Mono FIR Controller	1	1	FIR Controller blocks in Mono/Stereo with 1 to 5 ways. Each way has a 6- Band PEQ, a crossover with polarity, alignment delay, FIR filter, level control and a compressor/
			2	

			3	limiter. Importing factory-made Speaker Settings is possible. Exporting custom parameter sets as Speaker Settings is possible as well.
			4	
			5	
	n-Way Stereo FIR Controller	2	2	
			4	
			6	
			8	
			10	
Generators	Tone Generator	0	1	Tone Generator for generating a constant sine signal or sine signal sweep.
	Noise Generator	0	1	Noise Generator for generating white/pink noise.
Inputs/ Outputs	Analog Input	8	-	The Analog Line Input (AI-1) has 8 analog inputs. Each input has a fader, level meter, Mute and Invert buttons.
	Analog Output	-	8	The Analog Line Output (AO-1) has 8 analog outputs. Each output has a fader, level meter, Mute and Invert buttons.
	Digital Input	8	-	The Digital Input (DI-1) has 8 digital inputs. Each input has a fader, level meter, Mute and Invert buttons. For each receiver the input source can be selected and status and clock rate are shown.
	Digital Output	-	8	The Digital Output (DO-1) has 8 digital outputs. Each output has a fader, level meter, Mute and Invert buttons.
	Analog Microphone Input	8	-	The Analog Microphone Input (MI-1) has 8 analog inputs with sensitivities that are suitable for connecting microphones. Each input has a fader, level meter, gain control, Mute and Invert buttons. Activating phantom power for each input is also possible.
	CobraNet Input	8	-	The CobraNet Input (CM-1) has 8 input channels. Selecting a CobraNet bundle is possible by entering its number or the assigned name. Each channel has status LEDs, word length indication and a Mute button.
	CobraNet Output	-	8	The CobraNet Output (CM-1) has 8 output channels. Selecting a CobraNet bundle is possible by entering its number or the assigned name.
	Dante Input	8	-	The Dante Input (DM-1) has 8 input channels.
	Dante Output	-	8	The Dante Output (DM-1) has 8 output channels.
Loudspeaker Controllers	n-Way LS Controller	1	1	Loudspeaker Controller blocks in Mono/Stereo with 1 to 5 ways.
			2	
			3	

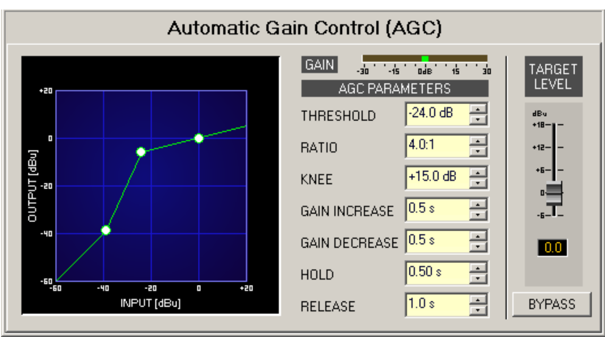
			4	Each way has a 6-Band PEQ, a crossover with polarity, alignment delay, level control and a compressor/limiter. Importing factory-made Speaker Settings for each Loudspeaker Controller way is possible. Exporting custom parameter sets as Speaker Settings is possible as well.
			5	
	n-Way Stereo LS Controller	2	2	
			4	
			6	
			8	
			10	
Matrix/Router	I x O Matrix Mixer	2...32	2...32	The Matrix Mixers have between 2 and 32 inputs and between 2 and 32 out- puts. Each input or output and each node of the matrix have individual level controls. An arbitrary amount of existing inputs can be mixed down to a single or several outputs.
	I x O Priority Matrix	2...32	2...32	The Priority Matrix has between 2 and 32 inputs and between 2 and 32 out- puts. Setting a priority for each input is possible. All outputs and all nodes of the matrix have individual level controls.
	I x O Matrix Router	2...32	2...32	The Matrix Router has between 2 and 32 inputs and between 2 and 32 out- puts. All inputs and outputs have individual level controls. Each input can be assigned to a single or several outputs. However, only one input can be active per output.
	Router	1...32	1...32	Router blocks with between 1 and 32 inputs and between 1 and 32 outputs. Each output can be fed by a single input source signal only. However, assigning an input to several outputs is possible.
Mixer	Mono Mixer	2..32	1	The Mono or Stereo Mixers have between 2 and 32 inputs and 1or 2 outputs. Each input has a fader, signal/clipping LEDs, Mute, Solo and Invert buttons while output channels have a fader, signal/clipping LEDs and a Mute button.
	Stereo Mixer	2...32	2	
	Automixer	2	3	The Auto Mixers have 2, 4, 6, 8, 10, 12, 16 or 24 inputs/ outputs and 1 additional auto mixed output. Each input has fader, meter, signal/clipping LEDs, Mute, Solo, Invert, Prio buttons and Auto checkbox while the auto mixed output has a fader, meter, Mute, Invert and Freeze Gain button. **HINT: For using the automixer block with 16 or 24 inputs a DSP-2DSP Extension Module is required.**
		4	5	
		6	7	
		8	9	
		10	11	
		12	13	
		16	17	
		24	25	
Utilities	Signal Meter	1	-	Signal Meter block indicating VU data.

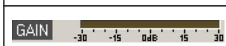
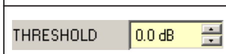
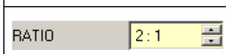
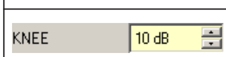
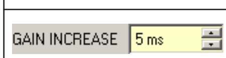
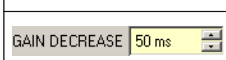
	Text Box			Editable Text box for labeling DSP configuration. Selectable font size, color and type.
	Superblocks			List of all available Superblocks.

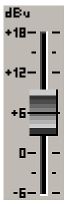
AUTOMATIC GAIN CONTROL (AGC)



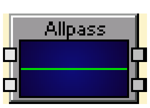
The DSP block Automatic Gain Control (AGC) manipulates the audio signal's dynamics. The AGC's task is to control a signal (mostly spoken word) to a defined level (average effective value, target level). This is accomplished by attenuating signals with levels that are above the target level and amplifying signals with levels that are below the target level. When compared to a compressor, control intervals are longer because level changes should not be obvious. An example for using the AGC in spoken word applications could be the automatic level adjustment at a panel discussion. In music applications the AGC can be used to automatically compensate level differences between various tracks or songs.



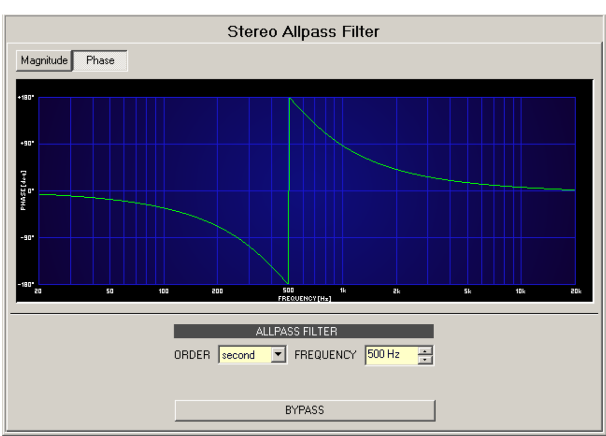
Element	Default	Range	Description
			This meter indicates by how much the AGC is attenuating or amplifying the signal level. A green box moving from the 0 dB mark to the left indicates level reduction. A green box moving from the 0 dB mark to the right indicates level amplification.
	-25 dBu	-30...0 dBu	THRESHOLD defines the level at which the AGC sets in. The THRESHOLD value always relates to the set TARGET value.
	5.0:1	1:1...15:1	RATIO defines the compression rate, i.e. to what degree the signal's level is reduced above the relative threshold level. A ration of 4.0 : 1 represents a signal reduction by factor 4.
	+10 dB	0.1... 15 dB	Settable bend of the compression curve below THRESHOLD. A moderate ratio prevents a drop/leap in amplification at the THRESHOLD.
	0.5 s	0.3... 20 s	GAIN INCREASE defines the time interval that it takes the AGC to raise the signal level to the TARGET.
	0.5 s	0.3...20 s	GAIN DECREASE defines the time interval that it takes the AGC to reduce the signal level to the TARGET.

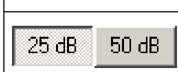
	0.05 s	0.05...60 s	HOLD defines how long the AGC keeps amplifying the signal after the signal level has dropped out of the AGC's operational range (below the set THRESHOLD and KNEE level, which are relative to TARGET).
	1 s	0.3...20 s	RELEASE defines the time that it takes to control the signal back to unity gain after the HOLD period has passed.
	0 dBu	-6...+18 dBu	TARGET fader for setting the desired output level (average RMS level) for the processed audio signal.
			This field shows the currently set TARGET as a numerical value. Entering the desired target value is possible as well.
			BYPASS activates (not engaged) or deactivates (engaged) the AGC, which allows quick A / B comparison between the original signal and the signal after AGC processing.

ALLPASS FILTER



When compared to other filter types, the All-Pass filter in the DSP block offers constant gain for all frequencies. However, All-Pass filters have frequency-dependent phase shifting (non-linear phase response), which is used for signal delay or phase equalization.



Element	**Default**	**Range**	**Description**
			Switch for selecting amplitude frequency response (magnitude) or phase response (phase) indication in the bode plot.
			Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)

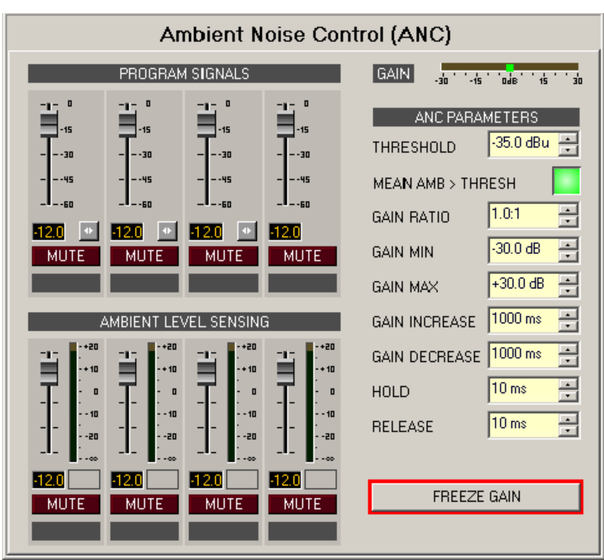
ORDER second	first	first, second	ORDER sets the desired order of the filter. A 1st order All-Pass filter shifts the phase by 180°. A 2nd order All-Pass filter shifts the phase by 360°.
FREQUENCY 500 Hz	1000 Hz	10 Hz...20 kHz	FREQ (frequency) sets the center frequency of the filter.
BYPASS			BYPASS activates (not engaged) or deactivates (engaged) the filter, which allows for quick A / B comparison between filtered and original sound signal.

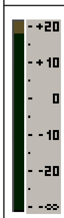
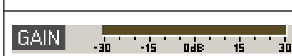
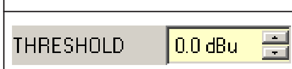
Editing filters via Mouse Dragging in the Graphic Display

If the filter has been activated (BYPASS is not engaged), a white dot in the frequency response graph represents the selected filter. Click onto this dot with the left mouse button and keep the button pressed down to change the filters frequency by dragging the mouse to the left or right.

AMBIENT NOISE CONTROL

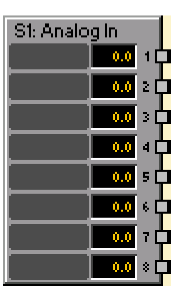
The DSP block Ambient Noise Control (ANC) automatically changes the level of up to 4 program signals depending on the actual ambient or background noise (ambient level), i.e. the ambient noise level in an area controls the volume setting, guaranteeing that the reproduction volume is always sufficient for announcements or alarm messages to be intelligible. Ambient noise can be picked up with up to 4 calibrated standard microphones.



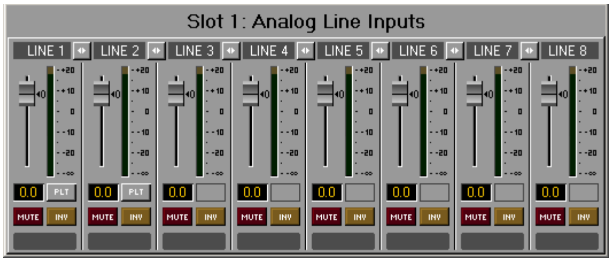
Element	Default	Range	Description
PROGRAM SIGNALS			
	-25 dBu	-30...0 dBu	Fader for setting the program signal level.
			This field shows the currently set numerical signal level value per fader. Entering the desired value is possible as well.
			LINK button groups neighboring input channels.
			MUTE button mutes the input signal.
			Text field for proving an input channel with an application specific name. **CAUTION: Using*(asterisk) and/or = (equal) signs in a name is not permissible.**
AMBIENT LEVEL SENSING			
	-25 dBu	-30...0 dBu	Fader for weighting different microphones (test signals) that are employed to register ambient noise.
			This field shows the currently set numerical signal level value per fader. Entering the desired value is possible as well.
			Shows the ambient noise signal’s RMS level.
			MUTE button mutes the input signal.
			Text field for proving an input channel with an application specific name. **CAUTION: Using*(asterisk) and/o r=(equal) signs in a name is not permissible.**
			This meter indicates the current degree of amplification/ attenuation applied to program signals.
	-35 dBu	-35...21 dBu	THRESHOLD defines the ambient noise level above which the program signal level is amplified.

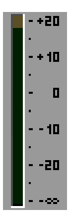
AMBIENT ABOVE THRESH			This indicator lights red when the current ambient noise level exceeds the THRESHOLD. It light green when the ambient noise level is below THRESHOLD.
GAIN RATIO 2 : 1	1.0:1	0.25:1...4:1	Indicates by how much the level of the program signal is amplified when the ambient noise level rises by 1 dB.
GAIN MIN 10 dB	-30 dB	-30... 30 dB	Defines the (minimum) level of the output signal when the ambient noise level is low (below THRESHOLD).
GAIN MAX +30.0 dB	+30 dB	-30...30 dB	Defines the maximum level of the output signal when the ambient noise level is high.
GAIN INCREASE 4s	1000 ms	10...3000 ms	Time constant for increasing the program signal level when the ambient noise level continuously exceeds the THRESHOLD. The time until the new signal level is reached is approx. 5-times the set response time.
GAIN DECREASE 0.5s	1000 ms	10...3000 ms	Time constant for decreasing the program signal level when the ambient noise level is continuously below the THRESHOLD.
HOLD 1s	10 ms	10...1000 ms	HOLD defines how long the program signal is maintained (amplified) when the ambient noise level falls below the THRESHOLD.
RELEASE 200 ms	10 ms	10...3000 ms	RELEASE defines how long it takes until the program signal level is controlled back to GAIN MIN after the HOLD time has passed.
FREEZE GAIN			Clicking on FREEZE GAIN memorizes the program signal's current gain. The is maintained even if the ambient noise level changes.

ANALOG LINE INPUTS

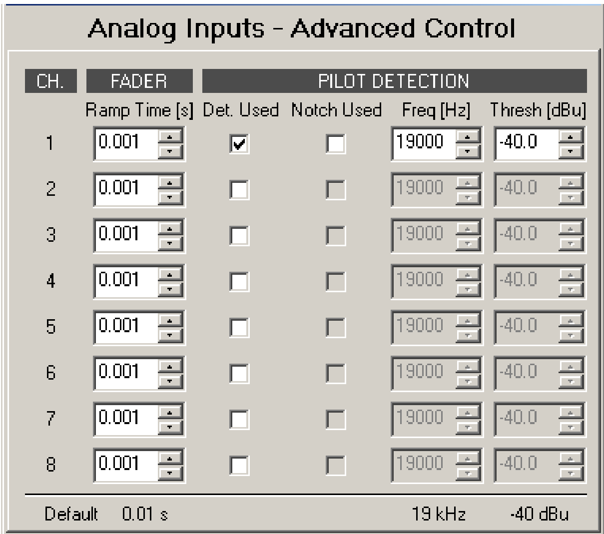


The Analog Line Inputs DSP block provides access to the eight analog inputs of an AI-1 Analog Input card. Establishing independent settings for each input channel is possible. A fader with ramping function as well as Mute and Invert buttons serve as controls. Each input channel has its individual level meter. Adjacent channels can be linked together using the LINK button. This allows for the convenient synchronous setting of several input channels, e.g. for stereo signals.
Each input channel has pilot tone detection, which, when activated, offers permanent monitoring of the output and the connecting audio cable of an external audio device. Activating and configuring pilot tone detection is possible in off-line mode from the Advanced Control window.



Element	Default	Range	Description
			Permanent channel labeling. Channels are numbered from LINE 1 to LINE 8.
			LINK button for linking (grouping) adjacent input channels.
	0.0 dB	-80...+18.0 dB	Fader for setting the input level. The Ramping Time that controls the fader’s ramping can be set in the Advanced Control window.
	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			Indicates the current input level.
			The PLT button activates (engaged) or deactivates (not engaged) pilot tone detection. The PLT button lights red when pilot tone detection is active but without a pilot signal being detected. With a pilot signal present, the PLT button lights green. The PLT button appears only when pilot tone detection has previously been activated in the Advanced Control window.
			MUTE button for muting the input signal.
			INV button for inverting the input signal’s polarity.
			Text field for labeling an input channel, e.g. giving it an application specific name. **CAUTION: Using*(asterisk) and/or = (equal) signs in a name is not permissible.**

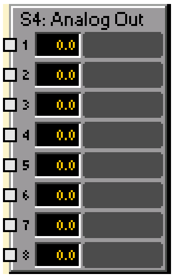
Click with the right mouse button on the DSP block and select Advanced Control from the pop-up context menu of the Analog Line Input block to open the Advanced Control window.



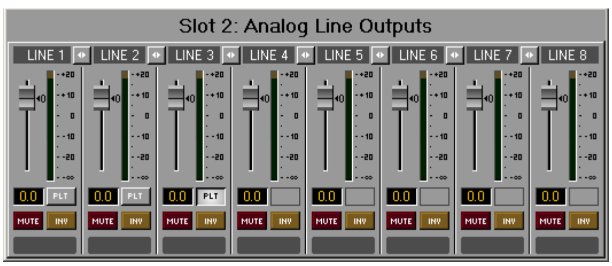
The following settings can be configured for each input channel of the Analog Line Input DSP block:

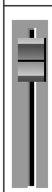
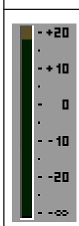
Element	Default	Range	Description
FADER Ramp Time [s] 0.001	0.001 s	0.001...20 s	A ramping time can be set for a channel's fader. When changing the signal level via Fader or Fader Display, within the specified period of time, the new signal level is set by means of the ramping function.
Det. Used			The checkbox activates the input's pilot tone detection. The PLT-Button only appears when pilot tone detection is active. **HINT: Det. Used can only be configured OFFLINE.**
Notch Used			The checkbox activates a notch filter when pilot tone detection is activated. The notch filter filters an existing pilot tone out of the input signal, so that it will not reach components that are connected behind the Analog Line Input. **HINT: Notch Used can only be configured OFFLINE.**
Freq [Hz] 19000	19000 Hz	20...20000 Hz	This field sets the frequency of the pilot signal to be detected. **HINT: Freq (Hz) can only be configured OFFLINE.**
Thresh [dBu] -40.0	-40 dBu	-60...0.0 dBu	This field sets the pilot tone detection's threshold. The analysis results in OK (green PLT button) when the level of the pilot signal exceeds the threshold. Without a pilot tone being present or if the signal level is below the set threshold, analysis results in a fault message on the corresponding input channel (red PLT button). **HINT: Thresh (dBu) can only be configured OFFLINE.**

ANALOG LINE OUTPUTS



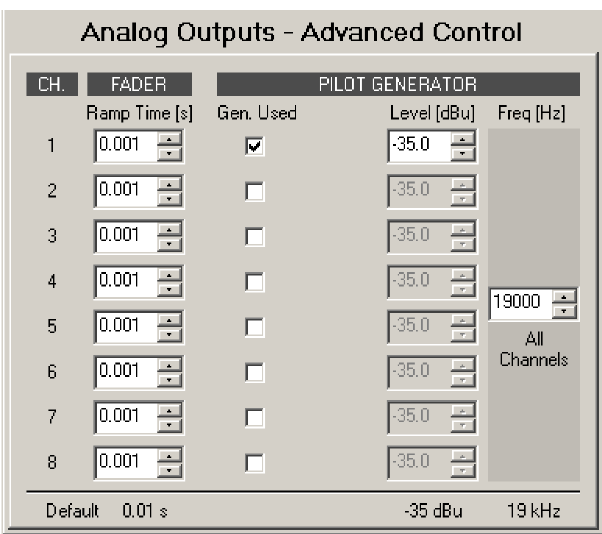
The DSP block Analog Line Outputs provides access to the eight outputs of an AO-1 Analog Output Card. Establishing independent settings for each output channel is possible. A fader with ramping function as well as Mute and Invert buttons serve as controls. Each output channel has its individual level indicator. Adjacent channels can be linked together using the LINK button. This allows for the convenient synchronous setting of several output channels, e.g. for stereo signals.
Each output channel has a pilot tone generator, which, when activated, allows permanent monitoring of the output and the connecting audio cable to an external audio device (e.g. amplifier) that has pilot detection. Activating and configuring the pilot tone signal is only possible in off-line mode from the Advanced Control window.



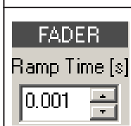
Element	Default	Range	Description
			Permanent channel labeling. Channels are numbered from LINE 1 to LINE 8.
			LINK button for linking (grouping) adjacent output channels.
	0.0 dB	-80...+18.0 dB	Fader for setting the output level. The Ramping Time that controls the fader's ramping can be set in the Advanced Control window.
	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			Indicates a channel's current output level.

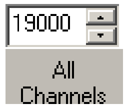
			The PLT button activates (engaged) or deactivates (not engaged) the pilot tone generator. The PLT button appears only when the pilot tone generator has previously been activated in the Advanced Control window.
			MUTE button for muting the output signal.
			INV button for inverting the output signal’s polarity.
			Text field for labeling an output channel, e.g. giving it an application specific name. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible**.

Click with the right mouse button on the DSP block and select Advanced Control from the pop-up context menu of the Analog Output block to open the Advanced Control window.

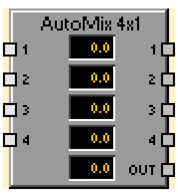


The following settings can be established for each output channel of the Analog Output DSP block:

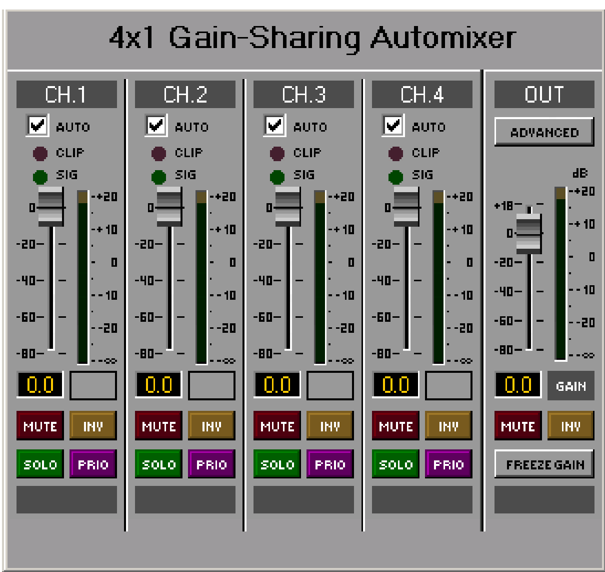
Element	Default	Range	Description
	0.001 s	0.001...20 s	A ramping time can be set for a channel’s fader. When changing the signal level via Fader or Fader Display, within the previously specified period of time, the new signal level is set by means of the ramping function.
			The checkbox activates the channel’s pilot tone generator. **HINT: Gen. Used can only be configured OFFLINE.**

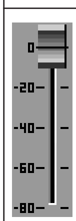
	-35.0 dBu	-60...0 dBu	This field allows setting the level of the pilot tone signal. **HINT: Level (dBu) can only be configured OFFLINE.**
	19000 Hz	20...20000 Hz	This field allows setting the frequency of the pilot tone signal. The set frequency applies to all outputs, for which the pilot tone signal has been activated. **HINT: Freq (Hz) can only be configured OFFLINE.**

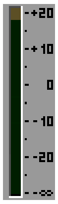
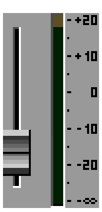
AUTO MIXER



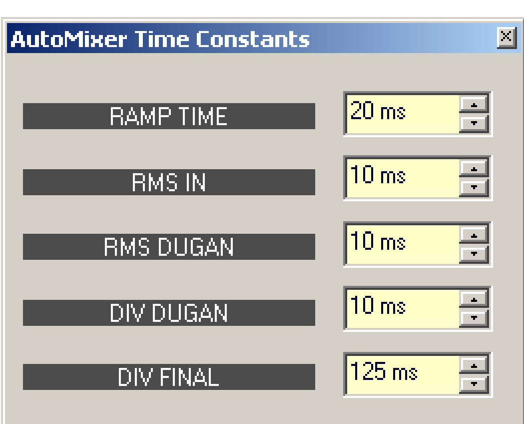
IRIS-Net provides Auto mixers with 2, 4, 6, 8, 10, 12, 16 or 24 inputs. For the 16 or 24 input Auto mixers a DSP-2 DSP extension card is required.



Element	Default	Range	Description
			Activates auto mixing function of corresponding input.
			The "CLIP" LED lights when the signal level nears clipping (+21dBu).
			If a signal is present at the input, the "SIG" LED lights.
	0.0 dB	-80...0 dB	(Post-)Fader for setting the levels in corresponding inputs. This fader works "post auto mixer algorithm", this means the level is changed after the automated evaluation of level distribution.

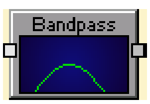
-12.0	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			Indicates the current input level. **CAUTION: The Meter is only active, if the check box AUTO of the input is selected.**
MUTE			MUTE button for muting the corresponding input signal.
INV			INV button for inverting the input channel.
SOLO			SOLO button for monitoring a single input signal.
PRIO			PRIO button for increasing priority (= higher level) of input signal.
			Text field for labeling an input channel with an internal IRIS-Net name. **CAUTION: The use of*(asterisk) and = (equal) in names is not permissible.**
ADVANCED			Opens the Time Constants dialog.
	0.0 dB	-80...+18.0 dB	Fader for setting the level of signal OUT. The meter indicates the current output level.
FREEZE GAIN			By pressing FREEZE GAIN the current input signal gain settings are stored and remain constant even if the level of the input signals changes.

Time Constants

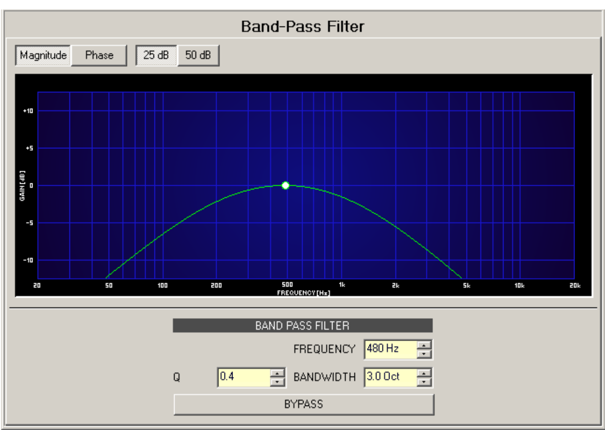


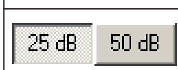
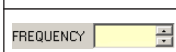
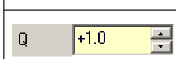
Element	Default	Range	Description
RAMP TIME	20 ms	1...20000 ms	Time constant of faders
RMS IN	10 ms	1...2000 ms	Time constant of RMS measurement of input signals.
RMS DUGAN	10 ms	1...2000 ms	Time constant of RMS measurement of dugan-gain weighted input signals.
DIV DUGAN	10 ms	1...2000 ms	Time constant for weighting rate of input signal level, based on total level.
DIV FINAL	10 ms	1...2000 ms	Time constant for rate of level change.

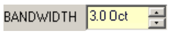
BAND-PASS FILTER



A band-pass filter is a device that passes frequencies within a certain range and rejects frequencies outside that range. The frequency and quality of the filter is adjustable.



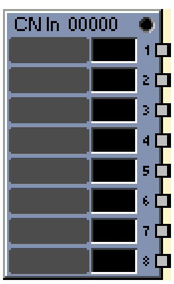
Element	Default	Range	Description
			Switches between frequency (magnitude) and phase response (phase) indication.
			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
	1000Hz	20 Hz...20 kHz	FREQUENCY sets the center frequency of the band-pass filter.
	1	0.4...100	The Q-value sets the quality and thus the response of the band-pass filter.

	1.4	0.1...3	BANDWITH sets the quality and thus the response of the band-pass filter.
			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the band-pass filter has on the sound.

Filter Editing via "Mouse Movement" in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking on this dot with the left mouse button and keeping the mouse button pressed down allows changing the selected filter's frequency by moving the mouse to the left or to the right. Clicking on the dot with the right mouse button and keeping the mouse button pressed down allows changing the Q-values of the filter.

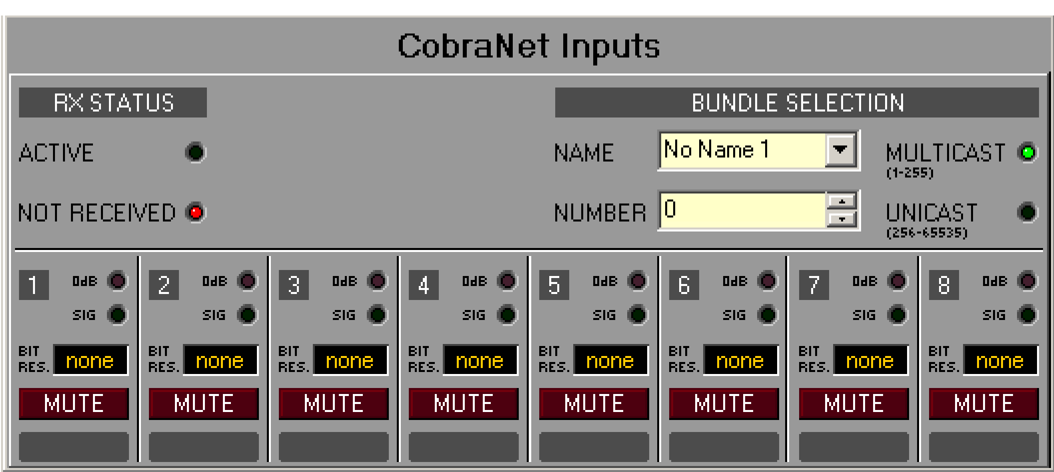
COBRA NET INPUTS

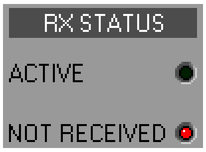


The DSP block CobraNet Inputs is part of the CM-1 CobraNet module. The dialogue is equivalent to a CobraNet bundle and provides 8 input channels from a CobraNet network. When on-line, the module's or the CobraNet bundle's RX-status is being indicated. Selecting a CobraNet bundle by its bundle number (or the assigned name) allows receiving up to eight pooled channels of that bundle. Signal status and resolution are individually indicated per channel, while separately muting each channel is possible as well.

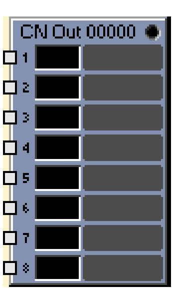
CobraNet differentiates between the following two kinds of bundles:

– Unicast Bundle: a Unicast Bundle (starting from bundle number 256) is transmitted to a single address, i.e. the bundle's destination is known.
– Multicast Bundle: a Multicast Bundle (bundle numbers 1 - 255) is not dedicated for one specific address. It is distributed throughout the entire network.

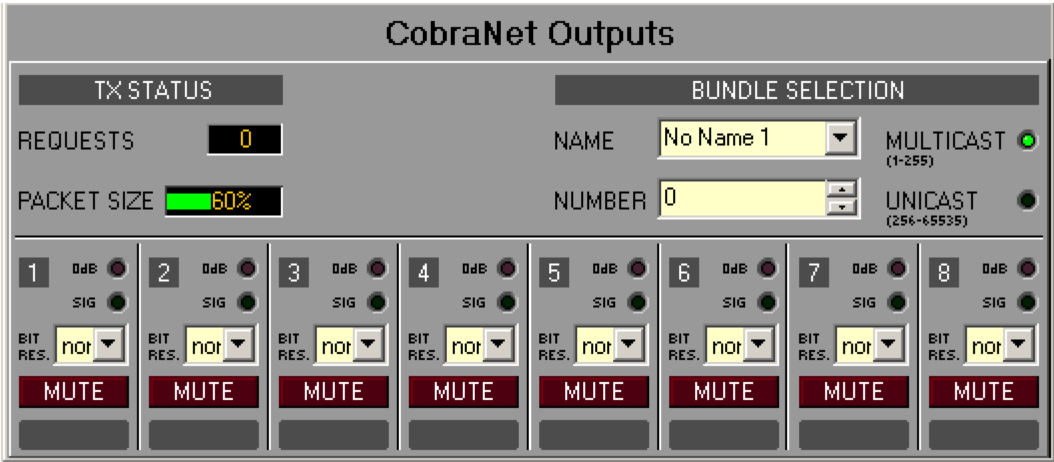


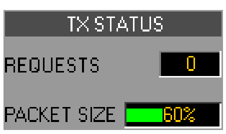
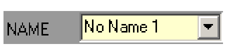
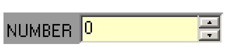
Element	Description
	The two LEDs in the RX STATUS area indicate the module’s current reception status. Whenever data is being received, the "ACTIVE" LED lights green. Whenever there is no data stream received via the CobraNet, the "NOT RECEIVED" LED lights red.
	Using alphanumeric characters in the name field allow the labeling of an incoming CobraNet bundle. For assigning identification to a bundle you first have to select its bundle number in the NUMBER field and then enter the desired name in the NAME field. Pressing the Return but- ton assigns the name to the selected bundle. Choosing a bundle with bundle number x which has no name assigned to yet, is possible by selecting "No Name x".
	The NUMBER field allows choosing the desired CobraNet bundle by selecting its bundle number.
	If the selected bundle is a Multicast Bundle (bundle numbers in the range between 1 and 255), the "MULTICAST" LED lights green. If the bundle number of the selected bundle is in the range of 256 up to 65535, the "UNICAST" LED lights green.
	The maximum amount of channels of the selected bundle is limited to eight. They are numbered from 1 to 8.
	The ”SIG” LED lights whenever a signal is being received on the channel of the selected bundle. The "0dB" LED light additionally, when the signal’s level is close to clipping.
	The field shows the transmitted bit rate (resolution) for the corresponding channel. Possible values are: 16, 20 and 24 Bit. The channels of a bundle can have different bit rates.
	The MUTE button allows muting the corresponding channel.
	Text field for assigning an exclusive IRIS-Net name to the corresponding channel. **CAUTION: Using * (asterisk) or = (equal) within a name is not permissible**.

COBRANET OUTPUTS

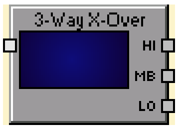


The DSP block CobraNet Outputs part of the CM-1 CobraNet module. The dialogue is equivalent to the one of a CobraNet bundle and offers up to 8 output channels. Send status of module or CobraNet bundle is indicated when on-line. When selecting a CobraNet bundle by Bundle Number (or assigned name) all channels combined in this bundle are present in the audio network. The signal status of each channel is indicated. Setting the word length (bit resolution) and individually muting each channel is possible as well.



Element	Description
	The REQUESTS field in the TX STATUS area shows the sending bundle’s number of recipients, if the (by bundle number or assigned name) selected bundle is a Unicast bundle. With Multicast bundles the amount of recipients is unknown and therefore not indicated. The PACKET SIZE field signals the actual load of the CobraNet in percent. The load depends on number and bit resolution of the channels included in the sending bundle.
	The Drop-Down field allows choosing a sending CobraNet bundle by selecting its assigned alphanumerical name. When assigning a name to a bundle, first select its bundle number in the NUMBER field and then enter the desired name in the NAME field. Pressing ”Return” actually assigns the name to the selected bundle. Choosing a bundle with bundle number x that has no name assigned yet is possible by selecting "No Name x".
	This field allows choosing the desired CobraNet bundle by selecting its bundle number.
	If the chosen bundle is a Multicast bundle (which is true for bundle numbers in the range between 1 to 255), the "MULTICAST" LED lights green. If the bundle number of the selected bundle is in the range between 256 and 65535, the "UNICAST" LED lights green.
	The up to eight possible channels of a selected bundle are numbered from 1 to 8.
	The "SIG" LED lights if a channel transmits a signal. In addition, the "0dB" LED lights whenever the level of the signal reaches the 0dB mark, signaling that there is a chance of driving the signal into distortion.
	The Drop-Down field allows specifying the desired data word length (bit resolution) for the corresponding channel. Possible values are16, 20 and 24 bit. When selecting none, no audio signal is being transmitted. Simultaneously grouping channels with different word length in a single bundle is possible.
	The MUTE button allows muting the corresponding channel.
	Text field for giving a channel an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

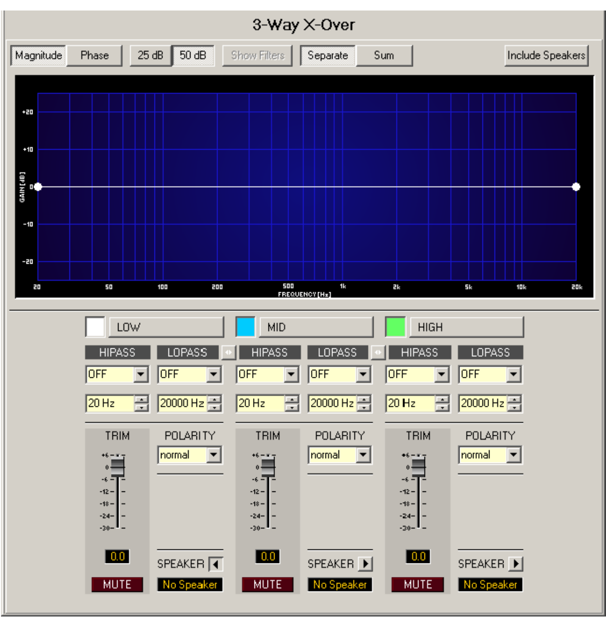
CROSSOVER (X-OVER)



A crossover (X-Over) allows splitting an input signal into various frequency bands to actively drive different types of loudspeakers from a power amplifier’s outputs. Crossovers with up to 5 ways in monaural (1 input) or stereo (2 inputs) quality are available. The following table lists the names of the respective outputs (frequency bands):

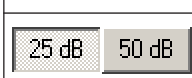
1 Way	FULLRANGE				
2Ways	LOW		HIGH		
3Ways	LOW	MID			HIGH
4Ways	SUB	LOW	MID		HIGH
5Ways	SUB	LOW	LOW-MID	HIGH-MID	HIGH

The X-Over window provides high and low pass filters, gain control and a polarity selection switch for each frequency band. Adjusting filter characteristics and matching levels per frequency band is possible through these parameters.



Graphic Display Indication

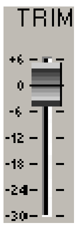
The crossover’s graphic display (Bode diagram) allows selecting several different graphic renditions which are listed in the following table:

Element	**Description**
	This switch lets you toggle between amplitude frequency response (Magnitude) and phase response (Phase) indication.
	This switch allows selecting an amplification axis scaling of 25 dB (± 12.5 dB) or of 50 dB (± 25 dB)

	Show Filters superimposes the individual filters' electrical transfer functions.
	Separate displays the transmission function of each frequency band separately. Sum shows an accumulated graph of all bands (ways) and the resulting transfer function of all filter and level settings at the outputs.
	This switch allows additional indication of measured loudspeaker transfer functions. This function takes effect only if speaker data has previously been loaded via Speaker.

Channel Parameters

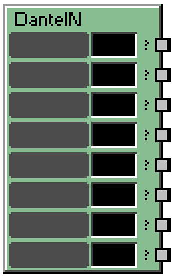
Element	Default	Range	Description
	thru, 20 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block is used for configuring the HIPASS filter. The user can choose from different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/oct. and 24 dB/oct. and cut-off frequencies of 20 Hz to 20 kHz.
	thru, 20000 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bes- sel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block is used for configuring the LOPASS filter. The user can choose from different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/oct. and 24 dB/oct. and cut-off frequencies of 20 Hz to 20 kHz.
			The LINK-Button allows linking (grouping) adjacent frequency bands (crossover ways). A frequency band’s LOPASS settings are automatically adopted to become the HIPASS settings of the adjacent band, and vice versa.

	0 dB	-30...+6 dB	The TRIM control allows raising the level of the selected frequency band's output signal by up to 6 dB or lowering it by as much as 30 dB. This allows for optimal level adjustment of individual frequency bands to match each other.
			This field indicates the currently set gain value numerically. However, manually entering the desired value is possible as well.
	normal	normal, inverted	POLARITY defines the polarity setting of the respective output. At times, inverting a channel's polarity (i.e. setting a negative polarity) might become necessary, depending on the crossover filter characteristic. The phase responses of some filter types can result in dropouts at the X-Over frequency when summing two channels. Inverting the polarity of the output carrying the lower frequency signal prevents this from happening. The following list shows all filter types that require inverting one channel of the corresponding pair of filters. – Bessel 12 dB – Butterworth 12 dB – Bessel 18 dB – Bessel 24 dB **HINT: The effect of the polarity parameter becomes very evident when the summed signal is displayed (set the switch to Sum).**
			MUTE-Button mutes the respective frequency band.
			The arrow button allows loading a speaker file. When Include Speakers has been activated, the frequency response shows the speaker transfer function included in the selected file.

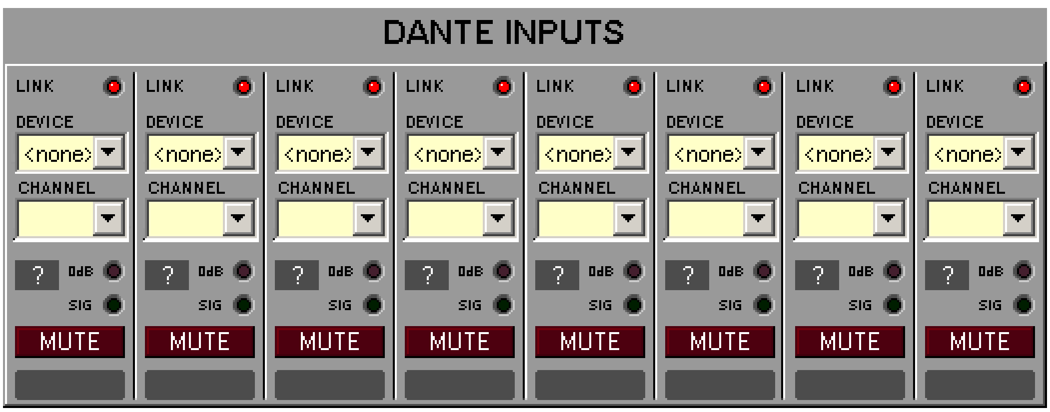
X-Over Filter Editing in the Graphic Display using the Mouse

With an X-Over filter being active (not set to OFF), the graphic display shows a colored spot on the amplitude frequency response. This spot represents the filter. Clicking with the left mouse button on this spot and holding the mouse button pressed down lets you adjust the frequency of the corresponding filter by dragging the spot to the left or to the right. The name of the respective filter is displayed in color when placing the mouse cursor over a spot.

DANTE INPUTS

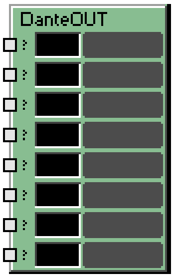


The Dante Inputs DSP block is part of the DM-1 Dante Interface module. The dialog provides 8 input channels from a Dante network.

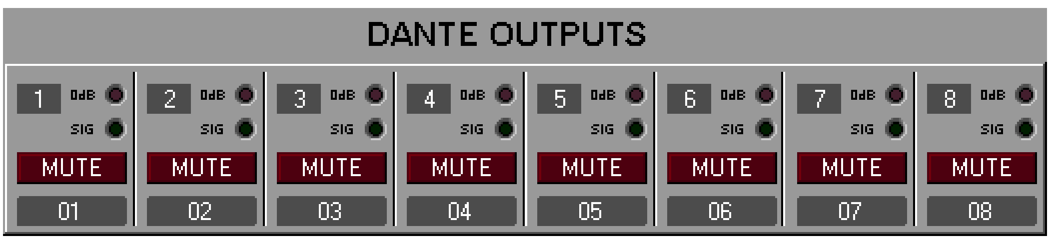


Element	Description
LINK	This LED indicates the input channel’s current reception status. Whenever data is being received, the LED lights green. Whenever there is no data stream received via Dante, the LED lights red.
DEVICE	The Drop-Down field allows selecting the name or description of the device the channel is received from.
CHANNEL	The Drop-Down field allows selecting the name or description of the received channel.
	The ”SIG” LED lights whenever a signal is being received on the channel. The "0dB" LED light additionally, when the signal’s level is close to clipping.
	The MUTE button allows muting the corresponding channel.
	Text field for assigning the Dante Channel Name to the corresponding channel. The name is also used in the Dante Controller. **CAUTION: Using * (asterisk) or = (equal) within a name is not permissible.**

DANTE OUTPUTS



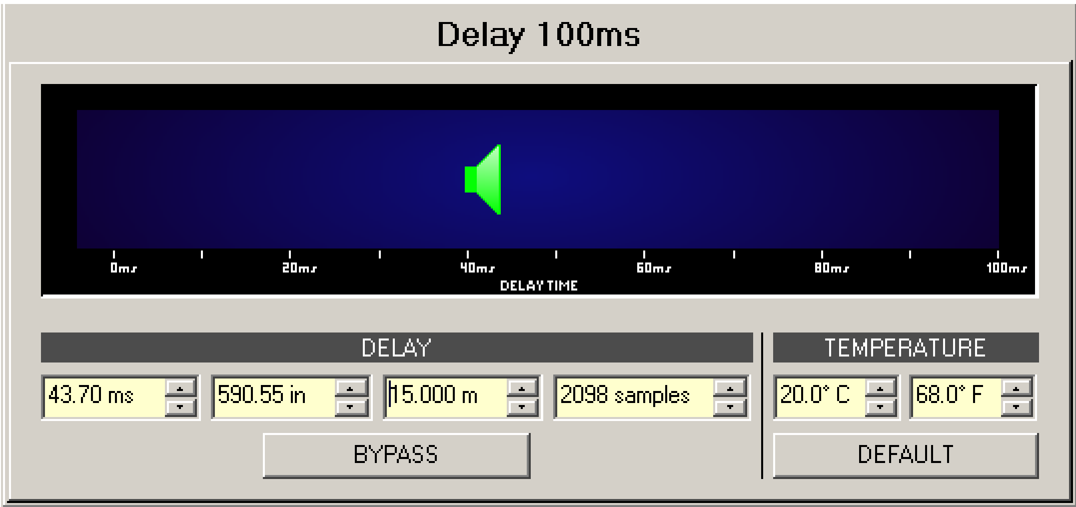
The Dante Outputs DSP block is part of the DM-1 Dante Interface module. The dialog offers up to 8 output channels. The signal status of each channel is indicated.

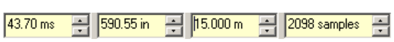


Element	Description
	The "SIG" LED lights if a channel transmits a signal. In addition, the "0dB" LED lights whenever the level of the signal reaches the 0dB mark, signaling that there is a chance of driving the signal into distortion.
	The MUTE button allows muting the corresponding channel.
	Text field for assigning the Dante Transmit Channel Name to the corresponding channel. The name is also used in the Dante Controller. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

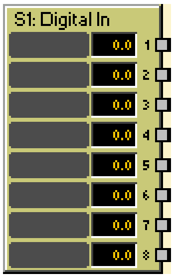
DELAY

The Delay DSP blocks allow the utilization of delay lines. This, for example, allows sending differently delayed audio signals to PA towers at open-air concerts to compensate for differences in distance among individual loudspeaker arrays. Alignment delay settings to compensate for individual transducers in a loudspeaker cabinet being mounted on different radiation planes is also possible.
Delay blocks in monaural quality (1 input and 1 output) and stereo quality (2 inputs and 2 outputs) are available. Both types allow for delay times of 10, 100, 500 or maximally 2000 ms.

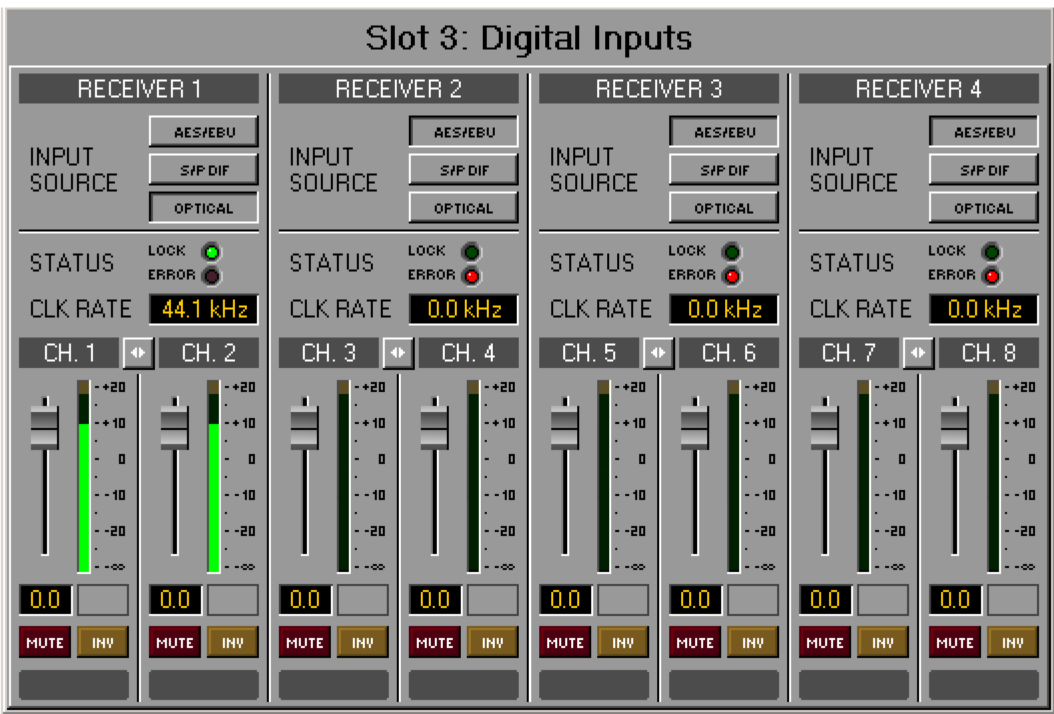


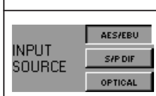
Element	**Description**
	Delay times can be entered directly in milliseconds (ms) or samples. Alternatively, entering a distance in inches (in) or meters (m) is possible as well. The appropriate delay time is calculated, taking into account the indicated Temperature.
	Temperature allows setting the current ambient temperature in degrees Celsius (° C) or degrees Fahrenheit (° F). The two units are automatically converted. The temperature parameter only takes effect if a distance value has previously been entered. In that case, the influence of the temperature is automatically taken into consideration during delay time calculation.
	BYPASS either activates (button not engaged) or deactivates (button engaged) the delay which allows for a quick A / B comparison of the effect that set parameters have on the sound characteristics.
	DEFAULT resets the temperature to 20° C or 68° F respectively.

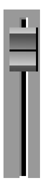
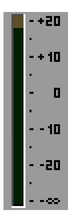
DIGITAL INPUTS



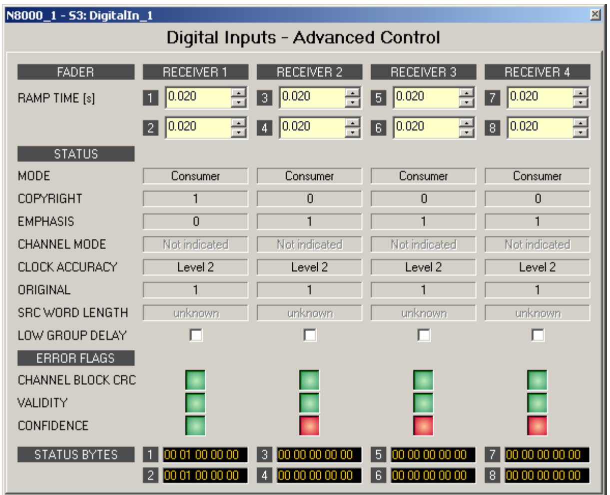
The Digital Inputs DSP block provides access to the eight digital inputs of an DI-1 Digital Input card. Establishing independent settings for each input channel is possible. A fader with ramping function as well as Mute and Invert buttons serve as controls. Each input channel has its individual level meter. Adjacent channels can be linked together using the LINK button. This allows simultaneous setting of several input channels, e.g. for stereo signals.



Element	Default	Range	Description
			The receivers are consecutively numbered from RECEIVER 1 to RECEIVER 4, offering two input channels, each.
			Switching the receiver's signal type. The DI-1 card's electrical inputs are capable of processing AES/EBU or S/P DIF compatible audio signals. Switching between the two formats has to be performed manually, using the appropriate labeled buttons. When using the optical input, select OPTICAL as the signal type.

			If the LOCK LED lights green, the input is synchronized to the incoming signal and the audio is correctly transmitted. When signal transmission fails, the ERROR LED lights red.
			Shows the sampling rate of the incoming signal when the input has been successfully synchronized. Sampling rates from 32 kHz to 192 kHz are supported.
			Permanent channel labeling. Channels are numbered from CH. 1 to CH. 8.
			LINK button for linking (grouping) adjacent input channels.
	0.0 dB	-80...+18.0 dB	Fader for setting the input level. The Ramping Time that controls the fader’s ramping can be set in the Advanced Control window.
	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			Indicates the current input level.
			MUTE button for muting the input signal.
			INV button for inverting the input signal’s polarity.
			Text field for labeling an input channel, e.g. giving it an application specific name. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

Click with the right mouse button on the DSP block and select Advanced Control from the pop-up context menu of the Digital Input block to open the Advanced Control window.

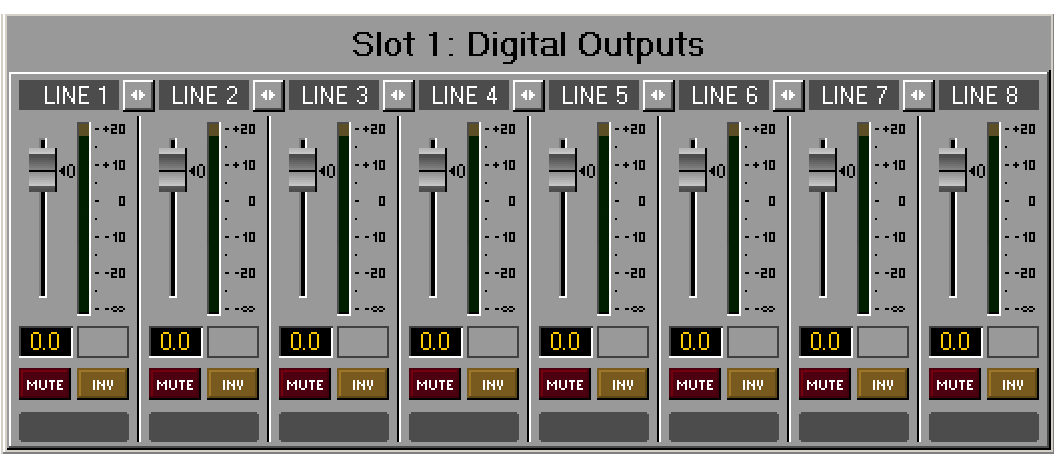


Element	Default	Range	Description
RAMP TIME [s] 1 0.020	0.020 s	0.001...20 s	A ramping time can be set for a channel's fader. When changing the signal level via Fader or Fader Display, within the specified period of time, the new signal level is set by means of the ramping function.
STATUS			
MODE		Consumer, Professional	Indicates the mode of the digital signal.
COPYRIGHT			Indicates if Copyright bit is set.
EMPHASIS			Indicates if Emphasis bit is set.
CHANNEL MODE		Not indicated, 2 channel, 1 channel, primary/ secondary, stereo, reserved for user applications, SCDSR, SCDSR (stereo left), SCDSR (stereo right), Multichannel	Indicates the Channel mode (in Professional mode only).

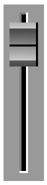
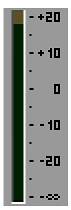
CLOCK ACCURACY		Level 1...3	Indicates the clock accuracy (in Consumer mode only)
ORIGINAL			Indicates if the Original bit is set.
SRC WORD LENGTH		not indicated, 17...24	Indicates the source word length (in Professional mode only)
LOW GROUP DELAY			Allow activating the option Low Group Delay of the sample rate converter‘s interpolation filter
ERROR FLAGS			
CHANNEL BLOCK CRC			Lights red, if the transmission of the Channel Status Block was not correct.
VALIDITY			Lights red, if the received audio data is not in PCM format.
CONFIDENCE			Lights red, if the signal quality is not good (e.g. Jitter)
STATUS BYTES 1 00 01 00 00 00			The first 5 bytes of the Channel Status block are indicated for each input channel.

DIGITAL OUTPUTS

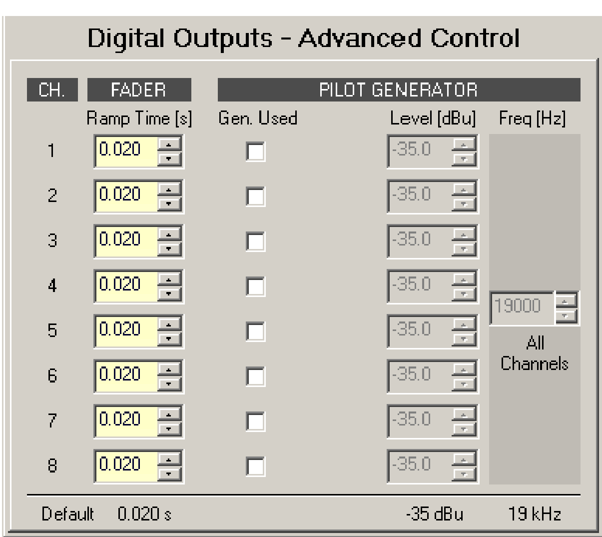
The DSP block Digital Outputs provides access to the eight outputs of an DO-1 Digital Output Card. Establishing independent settings for each output channel is possible. A fader with ramping function as well as Mute and Invert buttons serve as controls. Each output channel has its individual level indicator. Adjacent channels can be linked together using the LINK button. This allows for the convenient synchronous setting of several output channels, e.g. for stereo signals.

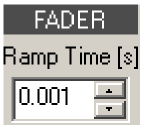


Element	Default	Range	Description
LINE 1			Permanent channel labeling. Channels are numbered from LINE 1 to LINE 8.

			LINK button for linking (grouping) adjacent output channels.
	0.0 dB	-80...+18.0 dB	Fader for setting the output level. The Ramping Time that controls the fader’s ramping can be set in the Advanced Control window.
	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			Indicates a channel’s current output level.
			MUTE button for muting the output signal.
			INV button for inverting the output signal’s polarity.
			Text field for labeling an output channel, e.g. giving it an application specific name. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

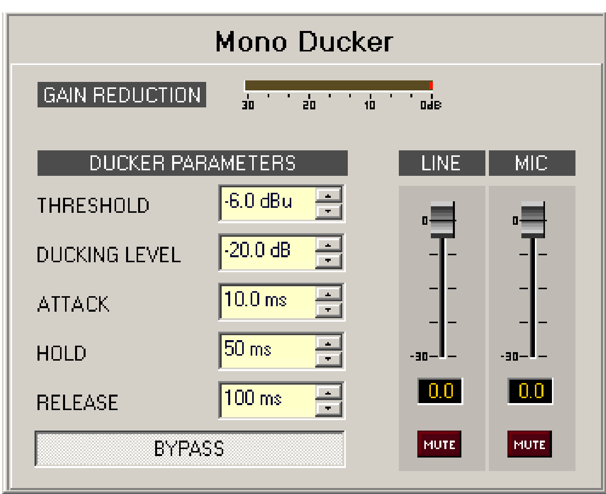
Click with the right mouse button on the DSP block and select Advanced Control from the appearing contextual menu of the Digital Output block to open the Advanced Control window.

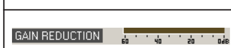


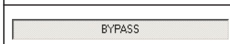
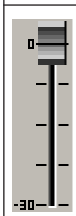
Element	Default	Range	Description
	0.001 s	0.001...20 s	A ramping time can be set for a channel's fader. When changing the signal level via Fader or Fader Display, within the previously specified period of time, the new signal level is set by means of the ramping function.
			The checkbox allows activating the channel's pilot tone generator. **HINT: Gen. Used can only be configured OFFLINE**.
	-35.0 dBu	-60...0 dBu	This field allows setting the level of the pilot tone signal. **HINT: Level (dBu) can only be configured OFFLINE.**
	19000 Hz	20...20000 Hz	This field allows setting the frequency of the pilot tone signal. The set frequency applies to all outputs, for which the pilot tone signal has been activated. **HINT: Freq (Hz) can only be configured OFFLINE.**

DUCKER

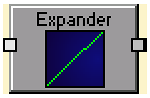
The Ducker DSP block reduces the level of a signal at the LINE input whenever a signal is present at the MIC input. If there is no MIC signal present the LINE signal automatically returns to its preset level.



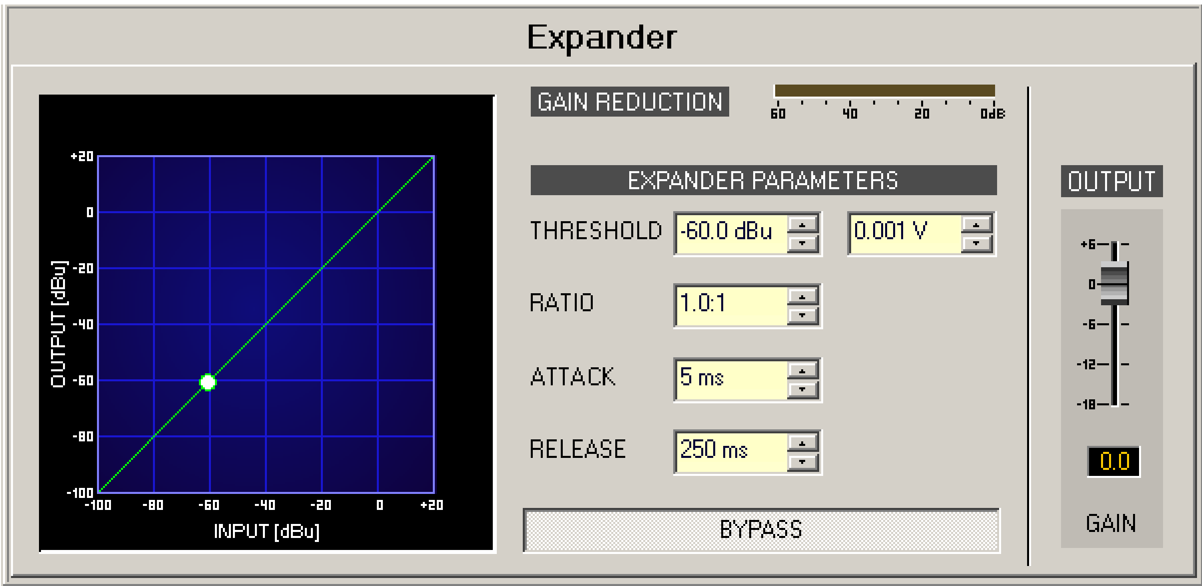
Element	Default	Range	Description
			This indicator signals in dB by how much the ducker reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.

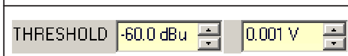
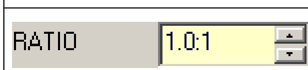
	-6.0 dBu	-15.0...+21.0 dBu	The THRESHOLD parameter defines the level value at which ducking sets in. The LINE signal will not be reduced as long as the signal level at the MIC input stays below the set threshold. As soon as the signal level at the MIC input reaches or exceeds the threshold, the LINE signal's level will be reduced by to the value set in DUCKING LEVEL.
	-20.0 dBu	-100.0...-6.0 dBu	The signal level of the LINE input is reduced by DUCKING LEVEL when the MIC signal level reaches or exceeds the THRESHOLD.
	10 ms	5...1000 ms	ATTACK defines how fast the gain of the LINE signal is reduced after the MIC signal exceeds the threshold level.
	50 ms	10...2000 ms	HOLD defines how long the level of the LINE input will continue to be reduced after the MIC signal drops below the set threshold.
	100 ms	5...1000 ms	RELEASE defines how fast the gain of the LINE signal is returned to the preset level once the MIC signal drops below the threshold level and the HOLD time has elapsed.
			BYPASS activates (not engaged) or deactivates (engaged) the ducker.
	0 dB	-30...0 dB	Fader for setting the MIC- or LINE-signal level at the ducker outputs. The setting of this faders has no effect on the ducking algorithm.
			The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			MUTE button for muting the MIC or LINE signal.

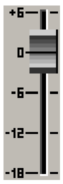
EXPANDER



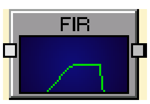
An Expander is used to reduce the level of a signal when it drops below a certain threshold resulting in an increase of the signal's overall dynamic range.



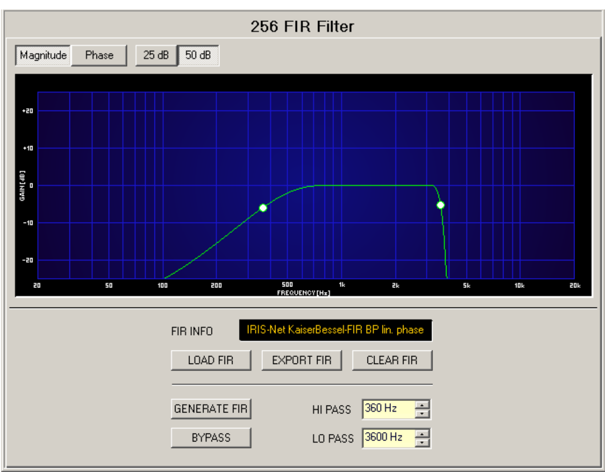
Element	Default	Range	Description
			This indicator signals in dB by how much the expander reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.
	-60.0 dBu or 0.001 V	-84.0...-25.0 dBu or 0.000...0.044 V	THRESHOLD defines the signal level above which the expander has no effect. If the signal level is below the THRESHOLD, the level is reduced according to the RATIO setting. Entering the desired value is possible in dBu as well as in V. The value can be entered in either box and will automatically be converted in the other.
	1.0:1	1.0:1...10.0:1	RATIO defines the compression rate, i.e. the degree of compression below the threshold level. For example, a ratio of 4.0 : 1 means the output signal will be reduced by 4 dB for every 1 dB the input signal drops below the threshold.
	5 ms	5...150 ms	ATTACK defines how fast the signal level will be reduced once it drops below the threshold. A short attack time means that even short signal dips below THRES- HOLD are efficiently expanded. Longer attack times leave signal dips below THRESHOLD untouched.

	100 ms	10...1000 ms	RELEASE defines how fast the output signal returns to the nominal level once the threshold level is exceeded.
			BYPASS activates (not engaged) or deactivates (engaged) the expander.
	0 dB	-30...0 dB	Fader for setting the output level.
			The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.

FIR FILTER

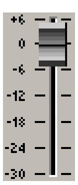


IRIS-Net provides FIR Filter blocks (Finite Impulse Response) of order 256 up to order 1792.

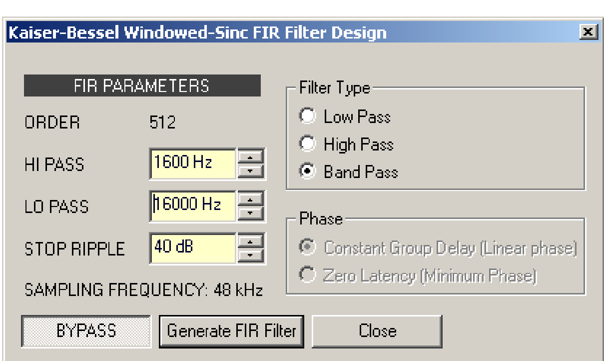


Channel Parameters

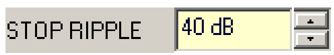
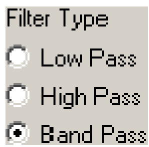
Element	Description
	Switch for displaying frequency response (magnitude) or phase response (phase)
	Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Description of the FIR filter currently in use.

LOAD FIR	After clicking onto LOAD FIR the "Open File..." dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT FIR	After clicking on EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click on the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
CLEAR FIR	Clears the current FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
GENERATE FIR	Clicking on the GENERATE FIR buttons opens the Filter Design dialog.
BYPASS	BYPASS switches the corresponding FIR filter ON (not engaged) or OFF (engaged), this provides a quick A / B comparison of the processed and unprocessed signal.
	Adjusts the gain of the signal between -30 dB and +6 dB.
HI PASS 200 Hz	HI PASS sets the cut-off frequency of the Hi pass filter.
LO PASS 2000 Hz	LO PASS sets the cut-off frequency of the Lo pass filter.

FIR Filter Design



Element	Default	Range	Description
ORDER 512			ORDER of the FIR filter.

	200 Hz	20...20000 ms	HI PASS sets the cut-off frequency of the Hi pass filter.
	2000 Hz	20...20000 ms	LO PASS sets the cut-off frequency of the Lo pass filter.
	40 dB	21...100 dB	STOP RIPPLE sets the slope of the FIR filter.
			Selects the FIR filter type of the DSP block.

Filter Editing via "Mouse Movement" in the Graphics Display

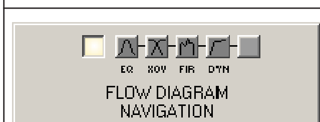
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter's frequency by moving the mouse to the left or to the right.

FIR/FIR-DRIVE CONTROLLER

The DSP block FIR Controller or FIR-Drive Controller entirely integrates the required signal processing for a complete PA system, providing FIR Controller or FIR-Drive Controller blocks starting from one way till up to four ways in monaural or stereo quality. The complete parameter set for a single loudspeaker is stored in the Speaker Setting file. IRIS-Net is shipped with a range of Speaker Setting files that hold
parameter sets which have been optimized for Electro-Voice and DYNACORD loudspeaker systems. An overview of the standard names for the different ways is presented in the following table:

1Way	FULLRANGE			
2Ways	LOW		HIGH	
3Ways	LOW	MID		HIGH
4Ways	SUB	LOW	MID	HIGH

Flow Diagram Navigation

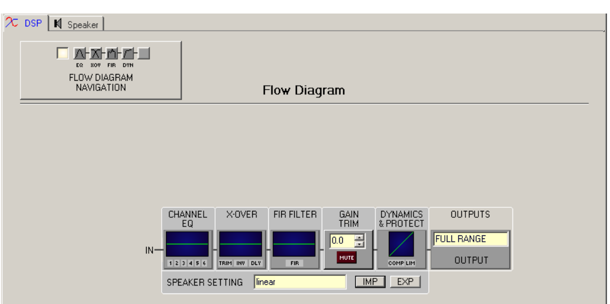
Element	**Description**
	The Flow Diagram Selector is available on all pages of the FIR Controller or FIR-Drive Controller block and allows navigating through the pages. Within the Flow Diagram Selector the user can select different DSP function blocks. The button for the currently selected block is pressed and shows a yellow background color.

A summary of the FIR Controller or FIR-Drive Controller processing blocks is provided in the following table. A more detailed description can be found in the corresponding paragraphs.

Element	FIR Controller (original)	FIR-Drive Controller
FLOW DIAGRAM	The signal flow chart offers an overview of the FIR Controller or FIR-Drive Controller blocks DSP settings. Also located in this area are all the controls for managing Speaker Settings.	
CHANNEL EQ	The CHANNEL EQ page provides access to the 6-Band parametric equalizers for loudspeaker equalization.	
X-OVER	The X-OVER page provides access to the crossover filters and parameters Gain, Polarity and Alignment delay for each way.	
FIR FILTER	This page provides access to the FIR filter.	
GAIN TRIM	TRIM allows matching the level of individual band passes while MUTE attenuates the corresponding band pass output.	
DYNAMICS & PROTECT	The DYNAMICS page provides access to a compressor and a Peak Anticipation limiter for each way.	The DYNAMICS page provides access to a Peak Anticipation limiter and a TEMP limiter for each way.

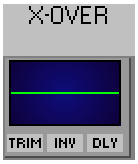
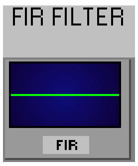
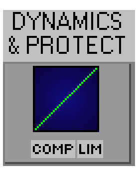
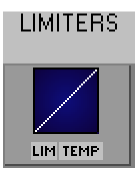
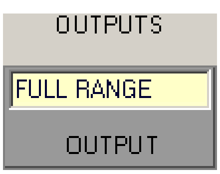
Flow Diagram

The FLOW DIAGRAM page shows a block diagram of the signal flow and provides a quick overview of all DSP settings. Channel labeling, muting and level adjustment can be carried out directly on the appropriate blocks. All other DSP parameters are accessible by clicking on the different function blocks. In addition, this window hosts all the controls for storing and loading Speaker Settings.
Selecting the FLOW DIAGRAM window is possible by clicking on the first or the last block in the Flow Diagram Selector.



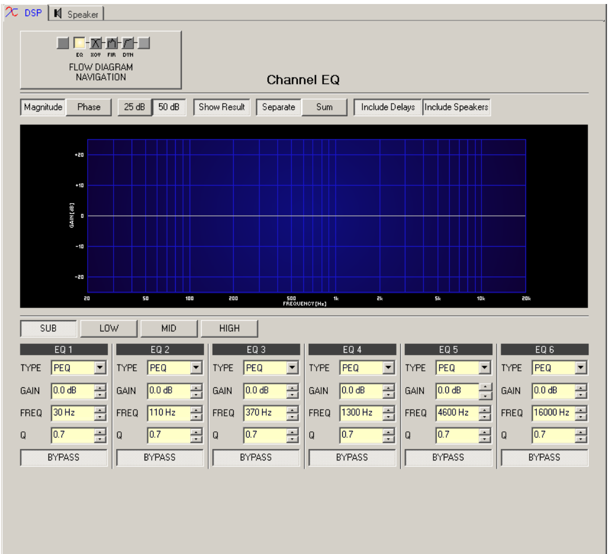
Function blocks

Element	Description
	This block represents the 6 Channel EQs of the FIR/FIR-Drive controller. The graph shows the frequency response of the Channel EQ Block. Clicking on this block opens the CHANNEL EQ page.

	This block represents the crossover of the FIR/FIR-Drive controller. The graph shows the crossover filters currently. Three additional LEDs show the state of trim, polarity and delay. Clicking on this block opens the X-OVER page.
	This block represents the FIR filter of the FIR/FIR-Drive controller.
	This block sets the level going into the Limiter stage. The setting is carried out via Up/Down Spin controls or by directly entering a level value in dB. The MUTE button is used to mute the signal. A left mouse click on the button will mute or unmute the signal.
 or 	This block represents the dynamics functions of the FIR/FIR-Drive controller. Two LEDs indicate whether compressor and/or limiter or Peak and/or TEMP limiter are activated. The graph indicates set values. Clicking on this block opens the DYNAMICS page.
	A name for the output can be entered in the text box.
	IMPORT opens a dialog box that allows loading Speaker Settings. This function imports a complete loudspeaker parameter set into the FIR/FIR-Drive controller. **CAUTION: A loaded Speaker Setting becomes immediately audible when in the on-line mode. Always make sure that the speakers setting that you are about to load is the desired speaker setting containing the correct parameter set. Otherwise, under most extreme circumstances, loading the wrong speaker set can permanently damage the connected loudspeaker systems!**
	EXPORT opens a dialog box that allows saving speaker settings. This function stores the complete settings of the FIR/FIR-Drive cont- roller as a parameter set in a file.

Channel EQ

The parametric 6-Band equalizer is used for loudspeaker equalization. Select the Channel EQ by clicking onto the second block in the Flow Diagram Selector or by double clicking on the CHANNEL EQ block in the large Signal flow Diagram.



Presentation in the Graphic Display

The following table summarizes the graphical controls shown in the display:

Element	Description
Magnitude Phase	Switch for selecting amplitude frequency response (magnitude) or phase response (phase) indication in the bode plot
25 dB 50 dB	Switch for scaling the magnitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings; the visible and audible result at the output. The audible result is displayed in bright colors while electrical graphs are indicated in dark colors.
Separate Sum	The Separate switch allows indicating the individual transmission function of the controller ways. The Sum switch allows indication of the summed signal of all ways.
Include Delays	Switch for including programmed delay in the frequency or phase response indication. The delays mainly affect phase response indication.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the Speaker register sheet.

Filter Parameter

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band.

TYPE Hipass	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, All- pass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope. – Allpass is a filter that has no influence on the frequency response but on the phase response in the transmission function.
SLOPE 12dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE defines the steepness or the filters order for Low-Shelving or High-Shelving filters as well as for Lo-Pass or Hi-Pass filters. Setting different slopes in the transmission range is possible. High-Pass filter and Q parameter together provide the possibility to program B6 alignments, which describes the raising in the cut-off frequency range.
FREQ 80 Hz	30 / 110 / 370 /	20 Hz...20	FREQ (frequency) sets the center frequency for parametric EQs or, in case of Shelving and High-Pass /
	1300 / 4600 /	kHz	Lo-Pass filters, it sets the cut-off frequency.
	16000 Hz		
Q +1.0	0.7	0.4...40.0	Q sets the quality or the bandwidth of the parametric EQ. A high Q-value results in a narrowband filter.
		(PEQ),	A low Q-value results in a wideband filter. Q also sets the quality and therefore the curve progression of
		0.4...2.0 (Hi-/	Lo-Pass-, High-Pass- and All-Pass filters with 12dB/Oct slope.
		Lopass),	
		0.4...2.0 (All-	
		pass)	
GAIN +2.5 dB	0 dB	-18...+12 dB	GAIN sets amplification (rising) or attenuation (lowering) of parametric EQs or Low-Shelving- and High-Shelving filters.
ORDER second	first	first, second	ORDER (All-Pass filters only) sets the desired order of an All-Pass filter. A 1st order All-Pass filter shifts the phase by 180°. A 2nd order All-Pass filter shifts the phase by 360°.
BYPASS			BYPASS activates (not engaged) or deactivates (engaged) the corresponding filter, which allows for quick A / B comparison between filtered and original sound signal.

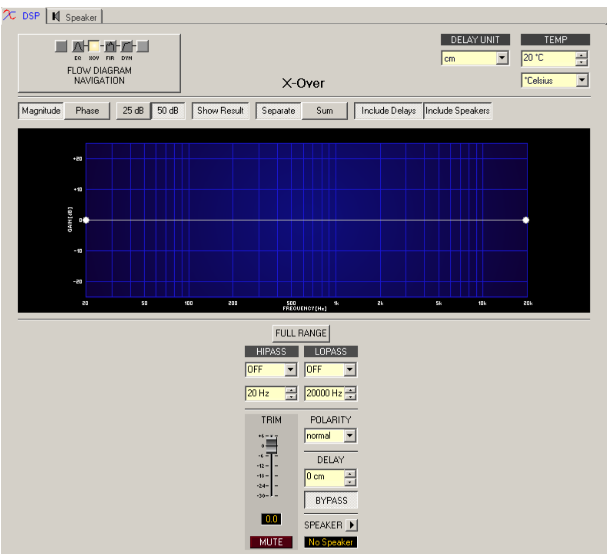
Editing filters via Mouse Dragging in the Graphic Display

If a filter has been activated (BYPASS is not engaged), a white dot in the frequency response graph represents the selected filter. Click onto this dot with the left mouse button and keep the button pressed down to change the corresponding filters frequency by dragging the mouse to the left or right. Moving the mouse up or down lets you set the filters gain (depending on the type of filter).

The name of a filter band is displayed in color whenever the mouse cursor is on top of its white dot, which improves clarity.

X-Over

The X-Over window provides a frequency crossover with Hi- and Lo-Pass filters, Delay, Gain-Trim and a polarity switch for each band pass of the FIR Controller or FIR-Drive Controller block. These parameters are used to separate the frequency bands of multi-way loudspeaker systems correctly, add delay to compensate for physical alignment of components in the loudspeaker and match the levels for each way.
Click on the third block in the Flow Diagram Selector or double click on the X-OVER block in the large Signal Flow Diagram to select the X-Over window.



Presentation in the Graphic Display

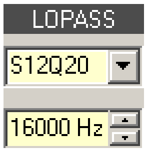
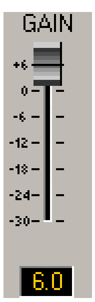
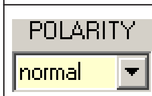
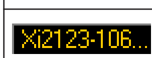
The following table summarizes the graphical controls shown in the display:

Element	Description
Magnitude Phase	Switch to select whether amplitude frequency response (magnitude) or phase response (phase) are being displayed.
25 dB 50 dB	Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB).
Show Result	Shows the resulting transmission function of all filter and level trim settings, which represents the visible and audible result. The resulting transmission function is indicated in light colors. The electric graphs are shown in dark colors.
Separate Sum	The Separate switch allows indicating the individual transmission function of the controller ways. The Sum switch allows indication of the summed signal of all ways.

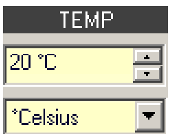
	Switch that allows including programmed delays in the frequency response or phase response indication. Include Delays primarily influences phase response indication.
	Switch to additionally display measured loudspeaker transmission functions. For this function to be effective you first have to load loudspeaker data sets under Speaker.

Filter Parameter

Element	**Default**	**Range**	**Description**
	OFF, 20 Hz	RESPONSE: OFF, 6dB, 12dB/Q=0.5, 12dB/ Q=0.6, 12dB/Q=0.7, 12dB/ Q=0.8, 12dB/Q=1.0, 12dB/ Q=1.2, 12dB/Q=1.5, 12dB/ Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bes- sel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HIPASS filter. Setting different filter types (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct and 24 dB/ Oct and cut-off frequencies between 20 Hz and 20 kHz is possible.

	OFF, 20000 Hz	RESPONSE: OFF, 6dB, 12dB/Q=0.5, 12dB/ Q=0.6, 12dB/Q=0.7, 12dB/ Q=0.8, 12dB/Q=1.0, 12dB/ Q=1.2, 12dB/Q=1.5, 12dB/ Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bes- sel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LOPASS filter. Setting different filter types (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct and 24 dB/ Oct and cut-off frequencies between 20 Hz and 20 kHz is possible.
	0 dB	-30...+6 dB	GAIN TRIM is for increasing the level of the corresponding channel by up to 6 dB or attenuating it by upto 30 dB, it is used to match the output levels for each fre- quency band.
	normal	normal, inverted	POLARITY allows inverting a channel, e.g. shifting its phase by 180°. Some cros- sover settings need phase inversion to prevent cancellations in the frequency response at the crossover frequency.
	0 cm	0...68643 cm	DELAY is used to delay the signal of the corresponding way. Typically, this delay is used as time alignment delay to overcome the difference in distance between the loudspeaker components within a cabinet.
			BYPASS activates (not engaged) or deactivates (engaged) the corresponding delay.
			The arrow sign next to SPEAKER opens a dialog for selecting speaker files.
			The text field shows the name of the currently loaded speaker file.

General Parameter

Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	Allows selecting the unit for delay settings.
	20 °C	-20...60 °C or -4...140 °F	Allows entering the ambient temperature. The temperature entered is taken into account by the calculations IRIS-Net uses when changing between different distance type delay units. Temperatures can be entered in °C or °F.

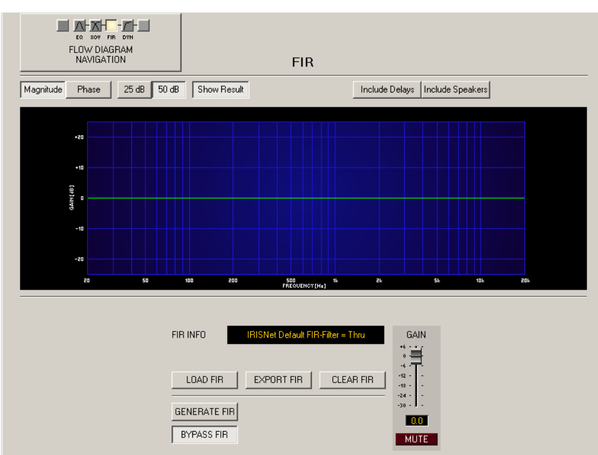
Editing X-Over filters via Mouse Dragging in the Graphic Display

If a X-Over filter has been activated (filter type not set to OFF), a white dot in the frequency response graph represents the filters cut-off frequency. Click onto this dot with the left mouse button and keep it pressed down to set the corresponding filters frequency by dragging the mouse to the left or right.

The name of a filter band is displayed in color whenever the mouse cursor is on top of its white dot, which improves clarity. In addition, another graph appears that represents the frequency response of the currently selected filter.

FIR Filter

Click on the fourth block in the Flow Diagram Selector or double click on the FIR block in the large Signal Flow Diagram to select the FIR Filter window.

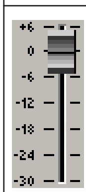


Graphics Display Indication

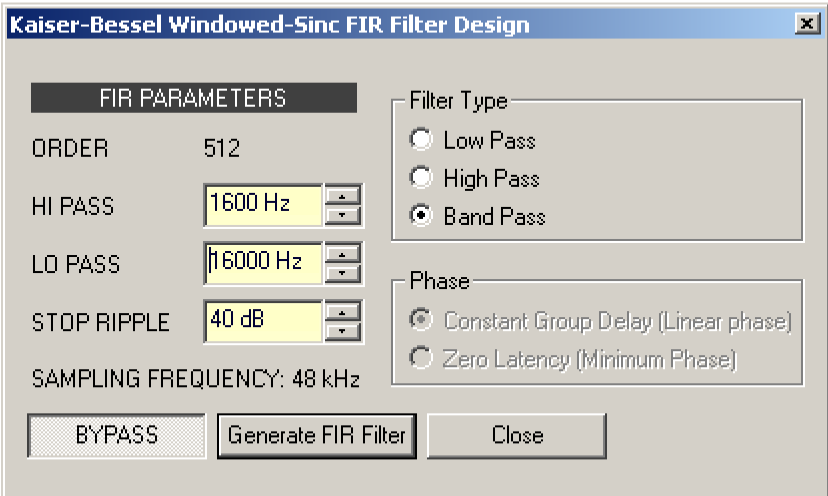
Element	Description
	Switch for displaying frequency response (magnitude) or phase response (phase)
	Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Displays the resulting transfer function of all filter and level trim settings, this is a visual representation of the audible result at the amplifier output. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.

Include Delays	This switch allows including the previously programmed delays in the frequency or phase response indication. The delays mainly affect the phase response.
Include Speakers	This switch is for the additional activation of actually measured speaker transfer functions to be included in the display. For this function to be effective, you have to load speaker data in the register ”Speaker” first.

Channel Parameters

Element	Description
FIR INFO IRISNet KaiserBessel-FIR BP lin. phase	Description of the FIR filter currently in use.
LOAD FIR	After clicking on LOAD FIR an "Open File..." dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT FIR	After clicking on EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click on the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
CLEAR FIR	Clears the current FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
GENERATE FIR	Clicking the GENERATE FIR buttons opens the Filter Design dialog.
BYPASS FIR	BYPASS switches the corresponding FIR filter ON (not engaged) or OFF (engaged), this provides a quick A / B comparison of the processed and unprocessed signal.
	Adjusts the gain of the signal between -30 dB and +6 dB.
0.0	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
MUTE	MUTE button for muting the output signal.

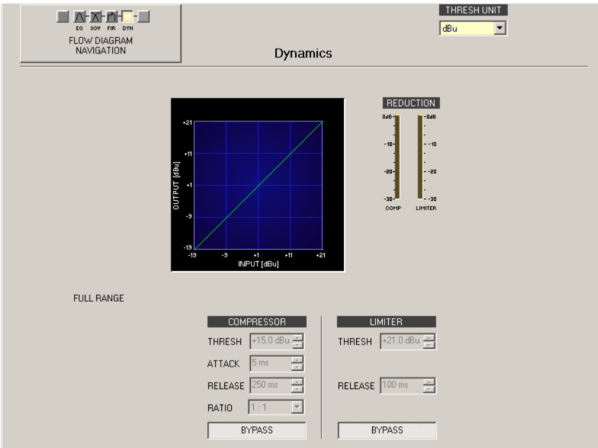
FIR Filter Design



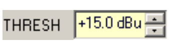
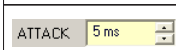
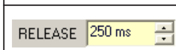
Element	Default	Range	Description
ORDER 512			ORDER of the FIR filter.
HI PASS 200 Hz	200 Hz	20...20000 ms	HI PASS sets the cut-off frequency of the Hi pass filter.
LO PASS 2000 Hz	2000 Hz	20...20000 ms	LO PASS sets the cut-off frequency of the Lo pass filter.
STOP RIPPLE 40 dB	40 dB	21...100 dB	STOP RIPPLE sets the slope of the FIR filter.
Filter Type Low Pass High Pass Band Pass			Selects the FIR filter type of the DSP block.

Filter Editing via ”Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right.

Dynamics (FIR Controller original)
The Dynamics window provides access to a compressor and a Peak Anticipation limiter. This allows setting the corresponding parameters in a band pass, so that connected loudspeaker systems are protected against dangerous level peaks and overload.
Click on the fifth block in the Flow Diagram Selector or double click on the DYNAMICS block in the large Flow Diagram to select the Dynamics window.



Compressor Parameter

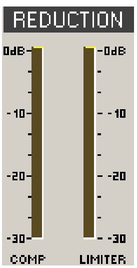
Element	Default	Range	Description
	15 dBu	-9.0...+21.0 dBu or 0.275... 8.696 V	THRESHOLD defines the signal level at which the compressor sets in.
	1 : 1	1 : 1, 1.4 : 1, 2 : 1, 4: 1, 8:1	RATIO defines the degree of compression above the threshold level. For example a ratio of 4.0 : 1 means the output signal will only increase by 1 dB for every 4 dB the input signal exceeds the threshold.
	5 ms	0...99 ms	ATTACK defines how fast the signal level will be reduced after it exceeds the threshold level.
	250 ms	50...999 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the compressor, which allows for quick A / B comparison between the compressed and uncompressed signals.

Limiter Parameters

Element	Default	Range	Description
	21 dBu	-9.0...+21.0 dBu or 0.275... 8.696 V	THRESHOLD defines the signal level at which the PA limiter sets in.

RELEASE 250 ms	250 ms	50...999 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
BYPASS			BYPASS activates (not engaged) or deactivates (engaged) the limiter, which allows for quick A / B comparison between limited and unlimited signals.

Meters

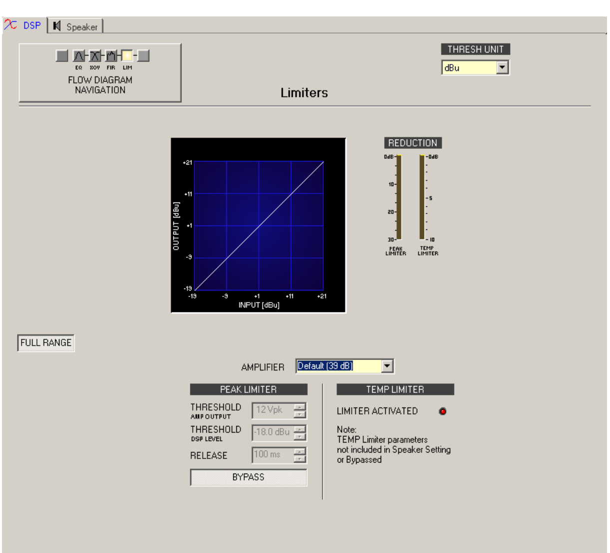
Element	Description
	These meters indicate signal reduction of the – compressor (COMP), – Peak limiter (LIMITER or PEAK LIMITER) or – TEMP limiter (TEMP LIMITER) in dB. A yellow bar increasing from top to the bottom indicates the degree of level reduction.

Editing Compressor / Limiter parameters via Mouse Dragging in the Graphic Display

If a compressor or limiter has been activated (Bypass is not engaged), the graphic display shows white dots that represent the according threshold values. Click onto one of these dots with the left mouse button and keep it pressed down to set the threshold value of the corresponding compressor or limiter by dragging the mouse up or down. Click with the right mouse button onto the compressors white dot and keep the mouse button pressed down to edit its ratio.

Limiters (FIR-Drive Controller)

The Limiters window provides access to a Peak limiter and a TEMP limiter. This allows setting the corresponding parameters so that connected loudspeaker systems are protected against dangerous level peaks and overload.
Click on the fifth block in the Flow Diagram Selector or double click on the LIMITERS block in the large Flow Diagram to select the Limiters window.

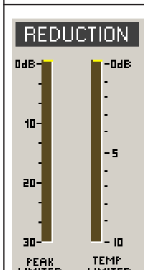


PEAK/TEMP-Limiter-Parameter

Element	Default	Range	Description
AMPLIFIER	Default (39 dB)	User, Default (39 dB), S900, S1200, CL800, CL1200, CL1600, CL2000, LX1600, LX2200, LX3000, L1000 (0dBu), L1000 (+6dBu), L1000 (26dB), L1600 (0dBu), L1600 (+6dBu), L1600 (26dB), L2400 (0dBu), L2400 (+6dBu), L2400 (26dB), H 2500 (0dBu), H 2500 (32dB), H 2500 (35dB), H 5000 (0dBu), H 5000 (32dB), H 5000 (35dB), SL900, SL1200, SL1800, SL2400, DSA8204, DSA8206, DSA8209, DSA8212, Q44, Q66, CP1200, CP1800, CP2200, CP3000S, CP4000S, P1200 (0dBu), P1200 (+6dBu), P1200 (26dB), P2000 (0dBu), P2000 (+6dBu), P2000 (26dB), P3000 (0dBu), P3000 (+6dBu), P3000 (26dB), TG5 (0dBu), TG5 (32dB), TG5 (35dB), TG7 (0dBu), TG7 (32dB), TG7 (35dB), Q44 MKII, Q66 MKII, Q99, Q1212, CPS2.4, CPS2.6, CPS2.9, CPS2.12	If the amplifier type used is not available in this dropdown, select the entry „User“. Then, select the field and enter the gain of the amplifier. Please refer to the technical documentation of the amplifier or ask your dealer for the correct gain setting.
THRESHOLD AMP OUTPUT	12 Vpk		THRESHOLD AMP OUTPUT determines the audio signal level above which the peak limiter starts operating. Please refer to the technical documentation of the used speaker for the power limit (program power or music power). then, use the Limiter Threshold Calculator (Tools > Limiter Threshold Calculator) to calculate the corresponding voltage at the amplifier output. The Speaker Settings for DYNACORD or Electro-Voice speakers already include the correct setting for this parameter.

THRESHOLD DSP LEVEL	-18.0 dBu		THRESHOLD DSP LEVEL determines the audio signal level above which the peak limiter starts operating. This value may change depending on the AMPLIFIER type that is selected, as the sensitivity and output power are automatically calculated with the Vpk value to provide DSP Threshold. The Speaker Settings for DYNACORD or Electro-Voice speakers already include the correct setting for this parameter.
RELEASE	100 ms	10 bis 999 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
BYPASS			BYPASS activates (not engaged) or deactivates (engaged) the corresponding limiter.
LIMITER ACTIVATED			The ACTIVATED LED lights green if the TEMP limiter is active. Most Speaker Settings for DYNACORD or Electro-Voice speakers already include the correct setting for the TEMP limiter, the LED lights green if settings are included.

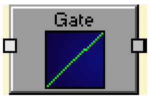
Meters

Element	**Description**
	These meters indicate signal reduction of the – Peak limiter (PEAK LIMITER) or – TEMP limiter (TEMP LIMITER) in dB. A yellow bar increasing from top to the bottom indicates the degree of level reduction.

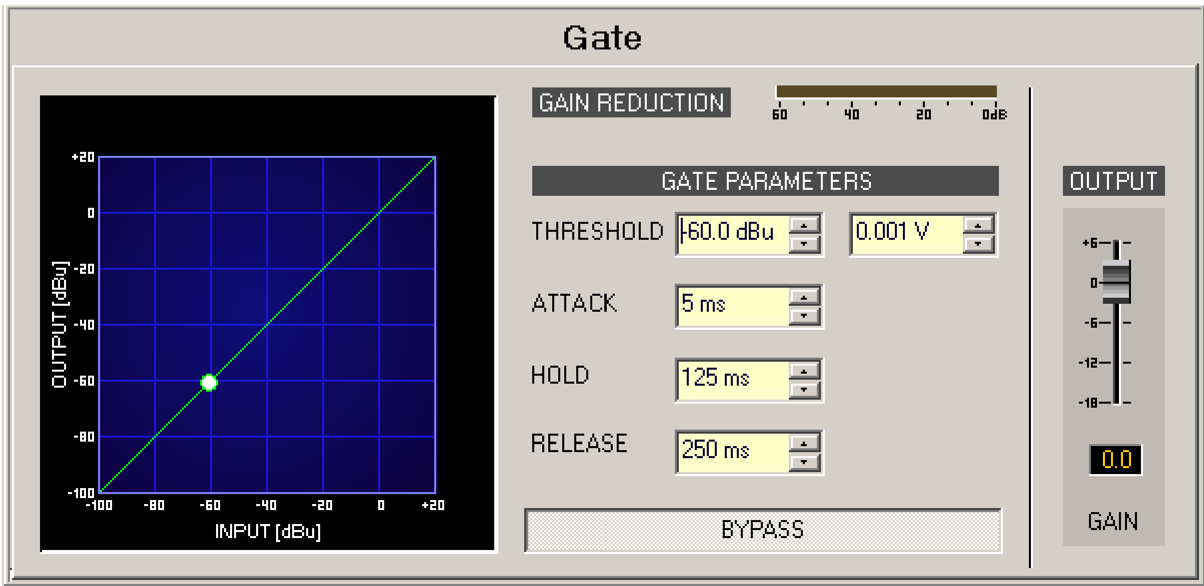
Editing Limiter parameters via Mouse Dragging in the Graphic Display

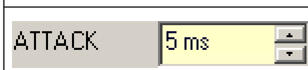
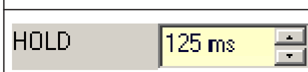
If the limiter has been activated (Bypass is not engaged), the graphic display shows a white dot that represents the according threshold value. Click onto one the dot with the left mouse button and keep it pressed down to set the threshold value of the compressor by dragging the mouse up or down.

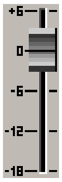
GATE



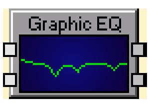
A Gate is used to mute a signal whenever its level falls below a specific threshold. This, for example, is useful to suppress unwanted noise of an open microphone channel.



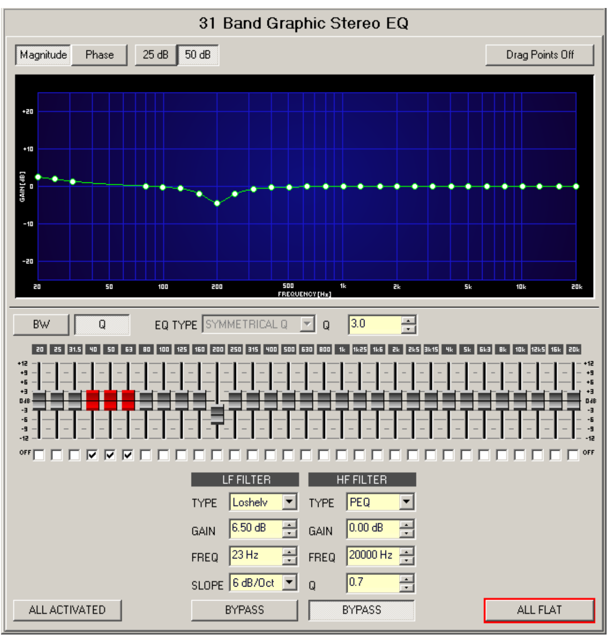
Element	Default	Range	Description
			This indicator signals in dB by how much the gate reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.
	-60.0 dBu or 0.001 V	-84.0...-25.0 dBu or 0.000...0.044 V	The THRESHOLD parameter defines the level value up to which the Gate mutes the signal. As long as the signal level is below the set threshold, the signal stays muted. As soon as the signal level at the input reaches or exceeds the threshold, the signal is transmitted. The signal's level, however, is not altered..
	5 ms	5...150 ms	The ATTACK parameter defines the speed at which the Gate opens. Short attack times result in the signal being transmitted, even if it exceeds the threshold only for a very short time.
	125 ms	5...1000 ms	HOLD defines how long the gate keeps transmitting the signal after the signal level has dropped below the threshold.
	250 ms	10...1000 ms	RELEASE defines the time that it takes to mute the signal after the HOLD period has passed.
			BYPASS activates (not engaged) or deactivates (engaged) the gate.

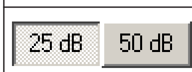
	0 dB	-18...6 dB	Fader for setting the output level.
			The fader display shows the numerical value of the current fader setting and also provides the possibility for entering a desired value.

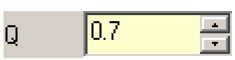
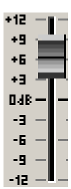
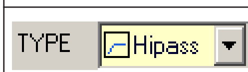
GRAPHIC EQUALIZER

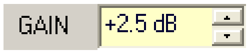
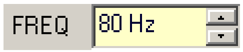
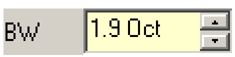
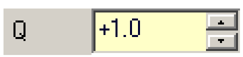


IRIS-Net provides 10, 15 and 31 band graphic equalizers in monaural and stereo quality.



Element	Default	Range	Description
			Switch to select magnitude or phase indication
			Switch to scale the amplification axis to 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
			Switch to select whether white points are visible during frequency response indication, or not.
			Switch to determine whether bandwidth BW or quality Q are selected when setting the current filter.

			Pressing ALL ACTIVATED deactivates all filters. The current status is stored when pres- sing ALL ACTIVATED while a single or several filters have been deactivated by marking the OFF option in the according checkbox. Pressing this button once again, e.g. activating the graphic equalizer, resets the formerly stored state.
	SYMMETRICA L Q	SYMMETRICAL Q, PROPORTIONA L Q, CONSTANT Q	Switching the graphic equalizer’s type between SYMMETRICAL Q, PROPORTIONAL Q and CONSTANT Q. SYMMETRICAL Q: The filters have an identical Q at all accentuation settings. The lowering frequency responses are symmetrical to the accentuation frequency responses. PROPORTIONAL Q: A Filter’s Q increases as soon as accentuation or lowering of the filter increases, with the effect that the equalizer becomes ”sharper” with increased EQ setting. The quality defined by Q corresponds to the quality at full accentuation or lowering. CONSTANT Q: The filter has the same Q at all accentuation or lowering settings. The resulting accentuation or lowering frequency response is not symmetrical.
	0.7	0.4...40.0	Q set the quality of all EQ bands. A high Q value results in a narrowband filter. A low Q value results in a wideband filter.
			The fixed frequencies of EQ bands
			Sets level amplification (accentuation) or reduction (lowering) of a band. A band’s fader is indicated in red when the band has been deactivated by marking the checkbox OFF. Pressing the spacebar resets the currently selected fader to 0 dB.
			Deactivating every single EQ-Band is possible by setting this checkbox. Deactivating a band does not change the band’s previously made settings.
			Freely programmable LF FILTER and HF FILTER are provided next to the 10, 15 or 31 bands.
	PEQ	PEQ. Loshelv. His- helv, Hipass, Lopass,	TYPE defines the desired LF FILTER or HF FILTER filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, quality and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. All pass is a filter which only affects the phase but not the frequency response of the transmission function.

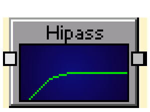
	0 dB	-18...+18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	20 / 20000 Hz	20 Hz...20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
 or 	1.9 Oct or 0.7	0.04...2.86 Oct or 0.4...40	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/ oct.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
			ALL FLAT resets all 10, 15 or 31 bands and also LF- and HF FILTER to 0 dB. **CAUTION: Using ALLFLAT discards all gain settings.**

Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS and Drag Points Off not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s amplification by moving the mouse up or down.

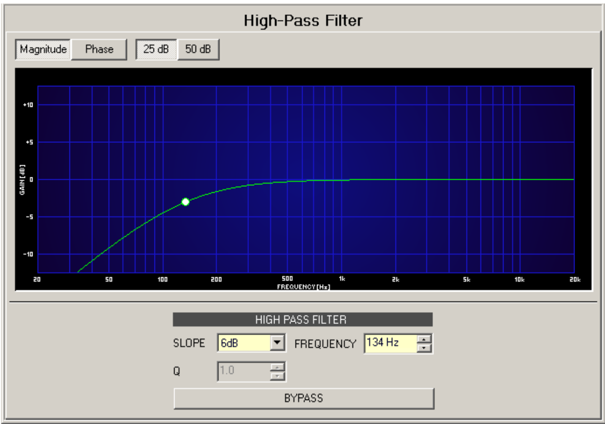
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

HIGH-PASS FILTER



High-pass filters pass high frequencies and stop low frequencies. Since it is not realistically possible to create a perfect filter that passes high frequencies totally unaltered and stops low frequencies completely, high-pass filter design involves compromises that allow some rounding of the corner at the filter cutoff frequency and some slope in the transition to the low frequency stop band. Different compromise schemes are given different names; examples are Bessel, Butterworth, and Linkwitz-Riley high-pass filter types.

The cutoff frequency is defined as the frequency at which the magnitude of the filter response has fallen to -3 dB relative to the unfiltered signal in the Bessel and Butterworth types, and to -6 dB in the Linkwitz-Riley types. The cutoff frequency is continuously variable from 10 Hz to 20 kHz.

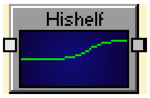


Element	Default	Range	Description
Magnitude Phase			Switches between frequency (magnitude) and phase response (phase) indication.
25 dB 50 dB			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
SLOPE 6dB	6dB	6dB, 12dB, 12dB BS, 18 dB BS, 24 dB BS (Bessel), 12dB BW, 18 dB BW, 24 dB BW (Butterworth), 12 dB LR, 24 dB LR (Linkwitz-Riley)	SLOPE sets the steepness or filter-order of the high-pass filter..
FREQUENCY	1000Hz	20 Hz...20 kHz	FREQUENCY sets the cut-off frequency of the high-pass filter.
Q +1.0	1	0.4...100	The Q-value sets the quality and thus the response of the high-pass filter with slope of 12dB/oct. This parameter is not available at other slopes.
BYPASS			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the high-pass filter has on the sound.

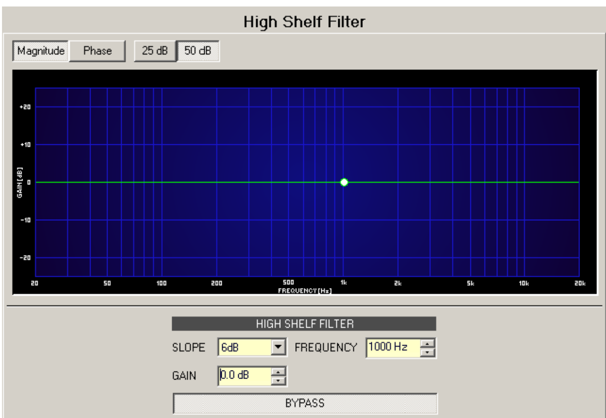
Filter Editing via ”Mouse Movement” in the Graphics Display

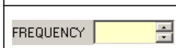
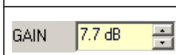
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the filter’s frequency by moving the mouse to the left or to the right. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values (if SLOPE is set to 12dB) of the filter.

HIGH SHELF FILTER



High shelf filters raise or lower the magnitude response at frequencies above the cut-off frequency without altering the response at frequencies below the cut-off frequency. Since it is not realistically possible to create a perfect filter that alters only high frequencies without affecting low frequencies, high shelf design involves compromises that allow some rounding of the corner at the filter cut-off frequency and some slope in the transition to the unaltered low frequencies.

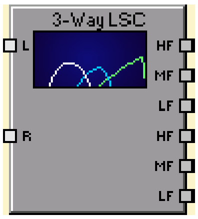


Element	Default	Range	Description
			Switches between frequency (magnitude) and phase response (phase) indication.
			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
	6dB	6dB, 12dB	SLOPE sets the transition band slope of the high shelf filter.
	1000Hz	20 Hz... 20 kHz	FREQUENCY sets the cut-off frequency of the high shelf filter.
	0.0 dB	-18...+18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of the high shelf filter.
			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the high shelf filter has on the sound.

Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the filter’s frequency by moving the mouse to the left or to the right.

LOUDSPEAKER CONTROLLER

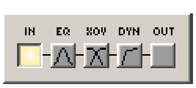


The DSP block Loudspeaker Controller entirely integrates the needed signal processing for a complete PA system, providing Loudspeaker Controller blocks starting from one way till up to five ways in monaural or stereo quality. An overview of the standard names for the different ways is presented in the following table:

1 Wege	FULLRANGE				
2 Wege	LOW		HIGH		
3 Wege	LOW	MID			HIGH
4 Wege	SUB	LOW	MID		HIGH
5 Wege	SUB	LOW	LOW-MID	HIGH-MID	HIGH

The complete parameter set for a single loudspeaker is stored in the Speaker Setting file. IRIS-Net is shipped with a range of Speaker Setting files that hold parameter sets which have been optimized for Electro-Voice and DYNACORD loudspeaker systems.

Flow Diagram Selector

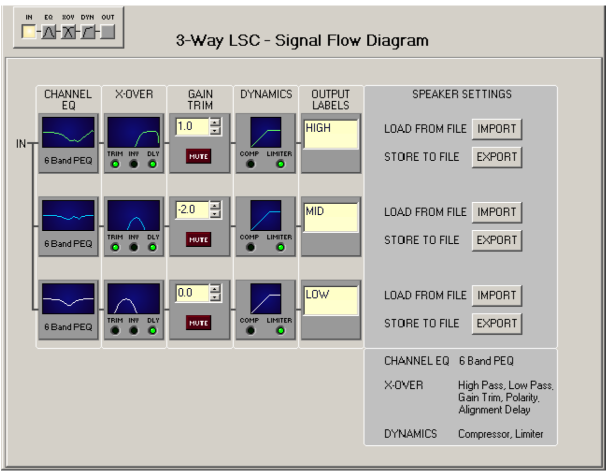
Element	Description
	The Flow Diagram Selector is available on all pages of the Loudspeaker Controller block and allows navigating through the pages. Within the Flow Diagram Selector the user can select different DSP function blocks. The currently selected block is indicated as engaged and in yellow.

Short descriptions of the Loudspeaker Controller blocks consecutive stages are to be found in the following table. A more detailed description can be found in the corresponding paragraphs.

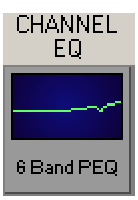
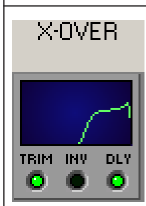
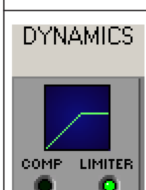
Element	Description
FLOW DIAGRAM	The signal flow chart offers an overview of the Loudspeaker Controller blocks DSP settings. Also located in this area are all the controls for managing Speaker Settings.
CHANNEL EQ	The CHANNEL EQ page provides access to the 6-Band parametric equalizers for loudspeaker equalizing.
X-OVER	The X-OVER area hosts the frequency crossover filters as well as parameters: Gain, Polarity and Alignment-Delay for all ways.
TRIM	TRIM allows matching the levels of individual ways, while MUTE attenuates the corresponding ways output.
DYNAMICS	The DYNAMICS page hosts a compressor and a limiter for each way.

Flow Diagram

The FLOW DIAGRAM window shows a signal flow diagram, which offers quick overview of all DSP settings. Separately muting, matching their levels amongst each other, and labeling channels is possible directly out of the diagram. All other DSP parameters are accessible via clicking on the different function blocks. In addition, this window hosts all necessary controls for storing and loading Speaker Settings.
Selecting the FLOW DIAGRAM window is possible by clicking onto the first or the fifth block in the Flow Diagram Selector.



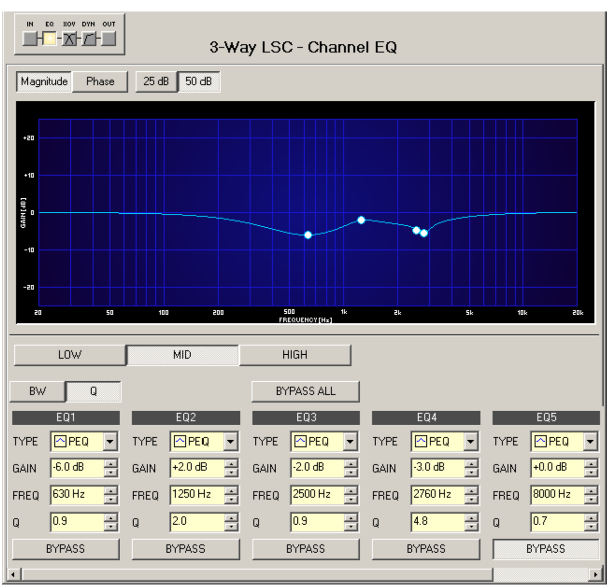
Function blocks

Element	Description
	This block represents the 6 Channel EQs of the individual ways. The graph indicates the frequency response of the Channel EQ Block. Simply clicking onto this block branches to the CHANNEL EQ page.
	This block represents the crossover of the corresponding way. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs signal the state of trim, polarity and delay. Simply clicking onto this block branches to the X-OVER page.
	This way allows separately setting the level. The setting is carried out via Up/Down Spin controls or by directly entering a level value in dB. The MUTE button is used to attenuate the corresponding way's signal. Clicking with the left mouse button on the MUTE button mutes the corresponding way. The MUTE button is shown engaged and lights red. Again clicking onto the button with the left mouse button deactivates the Mute function and the way is active.
	This block represents the dynamics functions of the corresponding way. Two LEDs indicate whether compressors or limiters are activated. The graph indicates set values. Simply clicking onto this block branches to the DYNAMICS page.

	The text field allows naming the corresponding way.
	IMPORT opens a dialog box that allows loading Speaker Settings. This function imports a complete loudspeaker parameter set into the corresponding way. **CAUTION: A loaded Speaker Setting becomes immediately audible when in the on-line mode. Always make sure that the speaker setting that you are about to load is the desired speaker setting containing the correct parameter set. Otherwise, under most extreme circumstances, loading the wrong speaker set can permanently damage the connected loudspeaker systems!**
	EXPORT opens a dialog box that allows saving speaker settings. This function stores the complete settings of the corresponding way as parameter set in a file.

Channel EQ

The parametric 6-Band equalizer is meant for loudspeaker equalization. Select the Channel EQ by clicking onto the second block in the Flow Diagram Selector or by double clicking onto the CHANNEL EQ block in the large Signal flow Diagram.



Presentation in the Graphic Display

The following table lists the graphic displays different graphic renditions:

Element	**Description**
	Switch for selecting amplitude frequency response (magnitude) or phase response (phase) indication
	Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)

Selecting the Way

Element	Description
LOW MID HIGH	Switch for selecting the way of the Loudspeaker Controller block for filter processing. The actual amount of switches depends on the type of Loudspeaker Controller block.

Filter Parameter

Element	Default	Range	Description
BW Q			Switch for selecting between bandwidth BW or quality Q when setting filters.
BYPASS ALL			BYPASS ALL deactivates all filters.
EQ 1			Name of the corresponding filter band.
TYPE Hipass	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the desired filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, quality and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 12dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE defines the steepness or the filters order for Low-Shelving or High-Shelving filters as well as for Lo-Pass or Hi-Pass filters. Setting different slopes in the transmission range is possible. High- Pass filter and Q parameter together provide the possibility to program B6 alignments, which describes the raising in the cut-off frequency range.
FREQ 80 Hz		20 Hz...20 kHz	FREQ (frequency) sets the center frequency for parametric EQs or, in case of Shelving and High- Pass / Lo-Pass filters, it sets the cut-off frequency.
Q +1.0	0.7	0.4...40.0 (PEQ), 0.4...2.0 (Hi-/ Lopass), 0.4...2.0 (Allpass)	Q sets the quality respectively the bandwidth of the parametric EQ. A high Q-value results in a narrowband filter. A low Q-value results in a wideband filter. Q also sets the quality and therefore the curve progression of Lo-Pass-, High-Pass- and All-Pass filters with 12dB/Oct slope.
GAIN +2.5 dB	0 dB	-18...+12 dB	GAIN sets amplification (rising) or attenuation (lowering) of parametric EQs or Low-Shelving- and High-Shelving filters.

ORDER second	first	first, second	ORDER (All-Pass filters only) sets the desired order of an All-Pass filter. A 1st order All-Pass filter shifts the phase by 180°. A 2nd order All-Pass filter shifts the phase by 360°.
BYPASS			BYPASS activates (not engaged) or deactivates (engaged) the corresponding filter, which allows for quick A / B comparison between filtered and original sound signal.

Editing filters via Mouse Dragging in the Graphic Display

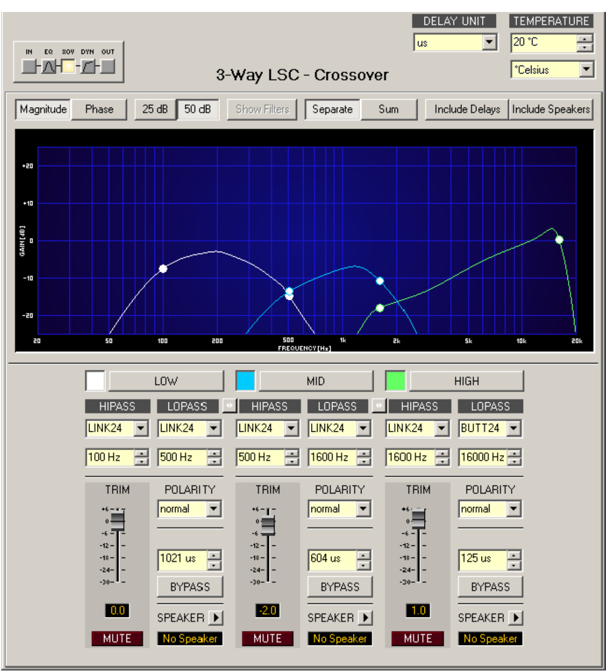
If a filter has been activated (BYPASS is not engaged), a white dot in the frequency response graph represents the selected filter. Click onto this dot with the left mouse button and keep the button pressed down to change the corresponding filters frequency by dragging the mouse to the left or right. Moving the mouse up or down lets you set the filters gain (depending on the type of filter).

The name of a filter band is displayed in color whenever the mouse cursor is on top of its white dot, which improves clarity.

X-Over

The X-Over window provides a frequency crossover with Hi- and Lo-Pass filters, Delay, Gain-Trim and a polarity switch for each Way of the Loudspeaker Controller block. These parameters allow separating the frequency bands of a multi-way loudspeaker system correctly, compensate natural delays and match levels.

Click on the third block in the Flow Diagram Selector or double click on the X-OVER block in the large Signal Flow Diagram to select the X-Over window.



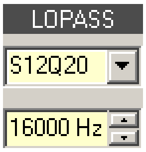
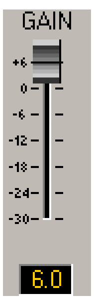
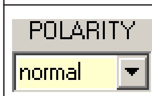
Presentation in the Graphic Display

The following table lists the graphic displays different graphic renditions:

Element	Description
Magnitude Phase	Switch to select whether amplitude frequency response (magnitude) or phase response (phase) are being displayed.
25 dB 50 dB	Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB).
Show Filters	Shows the resulting transmission function of all filter and level trim settings, which represents the visible and audible result. The resulting transmission function is indicated in light colors. The electric graphs are shown in dark colors.
Separate Sum	The Separate switch allows indicating the individual transmission function of the loudspeaker controller ways. The Sum switch allows indication of the summed signal of all ways.
Include Delays	Switch that allows including programmed delays in the frequency response or phase response indication. Include Delays primarily influences phase response indication. The influence of the delays have on the frequency response is more evident when indicating the summation.
Include Speakers	Switch to additionally display measured loudspeaker transmission functions. For this function to be effective you first have to load loudspeaker data sets under Speaker.

Channel Parameters

Element	Default	Range	Description
MID			Name of the corresponding way. For a loudspeaker controller block with 5 ways the standard names are: SUB, LOW, LOW-MID, HIGH-MID and HIGH.
HIPASS BUTT24 50 Hz	thru, 20 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HIPASS filter. Setting different filter types (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct and 24 dB/Oct and cut-off frequencies between 20 Hz and 20 kHz is possible.

	thru, 20000 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LOPASS filter. Setting different filter types (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct and 24 dB/Oct and cut-off frequencies between 20 Hz and 20 kHz is possible.
	0 dB	-30...+6 dB	GAIN TRIM is for amplifying the level of the corresponding channel by up to 6 dB or attenuating it by as much as 30 dB, which allows matching levels of the frequency bands amongst each other.
			The Mute button allows muting the signal of the loudspeaker cont- roller block's selected way.
	normal	normal, inverted	POLARITY allows inverting a channel, e.g. shifting its phase by 180°. Some frequency crossover settings need phase inversion, because otherwise, it dropouts appear at the crossover frequency. The influence of the polarity parameter becomes clearly evident in summed indication of the two amplifier channels (switch to Sum).
	0.0 ms	0.0...500.0 ms	DELAY allows delaying the signal of the corresponding way by a settable amount of time. Typically, this delay is used as time alignment delay to overcome the difference in distance between the loudspeaker systems within a cabinet.
			BYPASS activates (not engaged) or deactivates (enganged) the corresponding delay.

SPEAKER ▶			The arrow sign next to SPEAKER opens a dialog for selecting speaker files.
Xi2123-106...			The text field shows the name of the currently loaded speaker file.

General Parameters

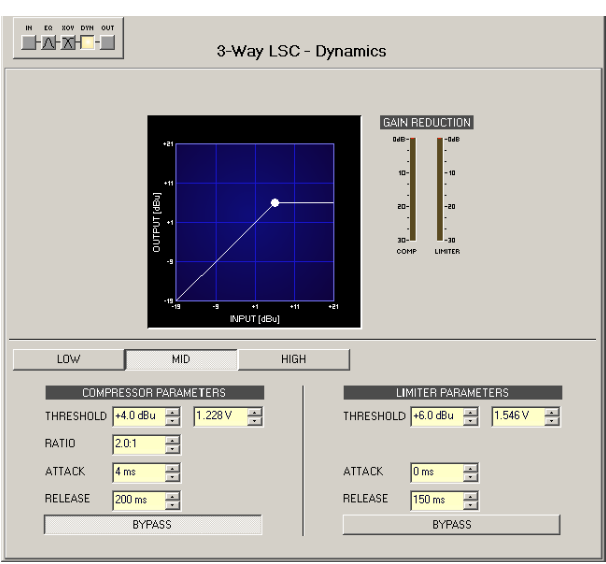
Element	Default	Range	Description
DELAY UNIT cm	ms	ms, samples, ft, in, m, cm, µs, s	Allows selecting the unit for delay settings.
TEMPERATURE +23 °C °Celsius	20 °C	-20...60 °C or -4...140 °F	Allows entering the ambient temperature. The entered temperature is taken into account to correct the actual delay time if the unit for distance has been specified for the delay. Temperatures can be entered in °C or °F.

Editing X-Over filters via Mouse Dragging in the Graphic Display

If a X-Over filter has been activated (filter type not set to OFF), a white dot in the frequency response graph represents the filters cut-off frequency. Click onto this dot with the left mouse button and keep it pressed down to set the corresponding filters frequency by dragging the mouse to the left or right.
The name of a filter band is displayed in color whenever the mouse cursor is on top of its white dot, which improves clarity. In addition, another graph appears that represents the frequency response of the currently selected filter.

Dynamics

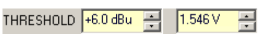
The Dynamics window provides access to a compressor and a limiter for each way. This allows setting the corresponding parameters in a way, so that connected loudspeaker systems are protected against dangerous level peaks and overload. Click on the fourth block in the Flow Diagram Selector or double click on the DYNAMICS block in the large Flow Diagram to select the Dynamics window.



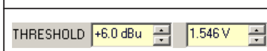
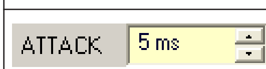
Selecting a Way

Element	Description
	Switch for selecting the loudspeaker controller blocks way for dynamic processing.

Compressor Parameters

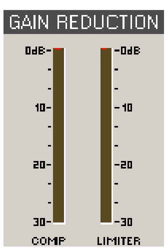
Element	Default	Range	Description
	21 dBu	-9.0...+21.0 dBu or 0.275... 8.696 V	THRESHOLD defines the signal level at which the compressor sets in.
	2 : 1	1: 1, 1.4: 1, 2: 1, 4 : 1, 8 : 1	RATIO defines the degree of compression above the threshold level. For example: A compression rate of 4 : 1 represents a signal reduction by factor 4.
	5 ms	0...99 ms	ATTACK defines the velocity the compressor sets in and reduces the gain when the threshold has been exceeded.
	250 ms	50...999 ms	RELEASE defines the time it takes for the compressor to return to normal level after the signal level declined the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the compressor, which allows for quick A / B comparison between the compressed and uncompressed signals.

Limiter Parameters

Element	Default	Range	Description
	21 dBu	-9.0...+21.0 dBu or 0.275... 8.696 V	THRESHOLD defines the signal level at which the limiter sets in.
	5 ms	0...99 ms	ATTACK defines how fast the gain is reduced after the signal level has declined the threshold level.

	250 ms	50...999 ms	RELEASE defines the time it takes for limiter to return to normal level after the signal level declined the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the limiter, which allows for quick A / B comparison between limited and unlimited signals.

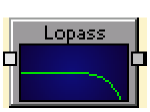
Meters

Element	Description
	These meters indicate signal reduction of the compressor (COMP) or limiter in dB. A yellow bar increasing from top to the bottom indicates the degree of level reduction.

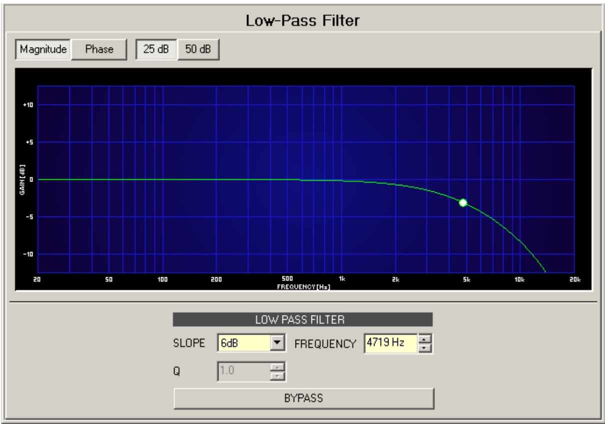
Editing Compressor / Limiter parameters via Mouse Dragging in the Graphic Display

If a compressor or limiter has been activated (Bypass is not engaged), the graphic display shows white dots that represent the according threshold values. Click onto one of these dots with the left mouse button and keep it pressed down to set the threshold value of the corresponding compressor or limiter by dragging the mouse up or down. Click with the right mouse button onto the compressors white dot and keep the mouse button pressed down to edit its ratio.

LOW-PASS FILTER



Low-pass filters pass low frequencies and stop high frequencies. Since it is not realistically possible to create a perfect filter that passes low frequencies totally unaltered and stops high frequencies completely, low-pass filter design involves compromises that allow some rounding of the corner at the filter cutoff frequency and some slope in the transition to the high frequency stop band. Different compromise schemes are given different names; examples are Bessel, Butterworth and Linkwith-Riley lowpass filter types.
The cutoff frequency is defined as the frequency at which the magnitude of the filter response has fallen to -3 dB relative to the unfiltered signal in the Bessel and Butterworth types, and to -6 dB in the Linkwitz-Riley types. The cutoff frequency is continuously variable from 20 Hz to 20 kHz.

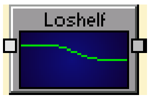


Element	Default	Range	Description
Magnitude Phase			Switches between frequency (magnitude) and phase response (phase) indication.
25 dB 50 dB			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
SLOPE 6dB	6dB	6dB, 12dB, 12dB BS, 18 dB BS, 24 dB BS (Bessel), 12dB BW, 18 dB BW, 24 dB BW (Butter- worth), 12 dB LR, 24 dB LR (Linkwitz-Riley)	SLOPE sets the steepness or filter-order of the low-pass filter.
FREQUENCY	1000Hz	20 Hz...20 kHz	FREQUENCY sets the cut-off frequency of the low-pass filter.
Q +1.0	1	0.4...100	The Q-value sets the quality and thus the response of the low-pass filter with slope of 12dB/oct. This parameter is not available at other slopes.
BYPASS			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the low-pass filter has on the sound.

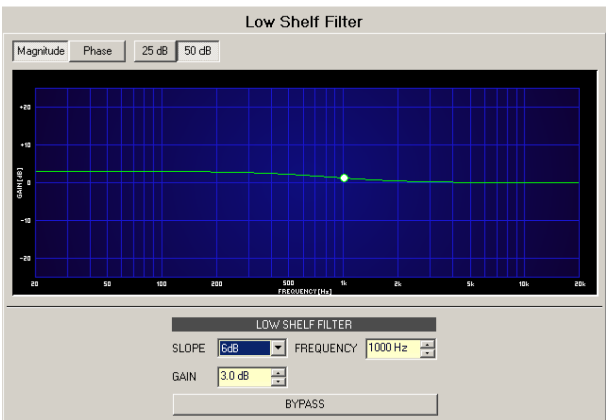
Filter Editing via ”Mouse Movement” in the Graphics Display

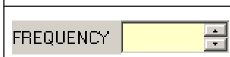
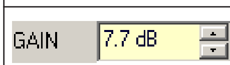
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the filter’s frequency by moving the mouse to the left or to the right. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values (if SLOPE is set to 12dB) of the filter.

LOW SHELF FILTER



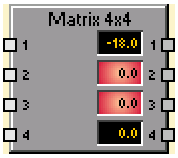
Low shelf filters raise or lower the magnitude response at frequencies below the cut-off frequency without altering the response at frequencies above the cut-off frequency. Since it is not realistically possible to create a perfect filter that alters only low frequencies without affecting high frequencies, low shelf filter design involves compromises that allow some rounding of the corner at the filter cut-off frequency and some slope in the transition to the unaltered high frequencies.



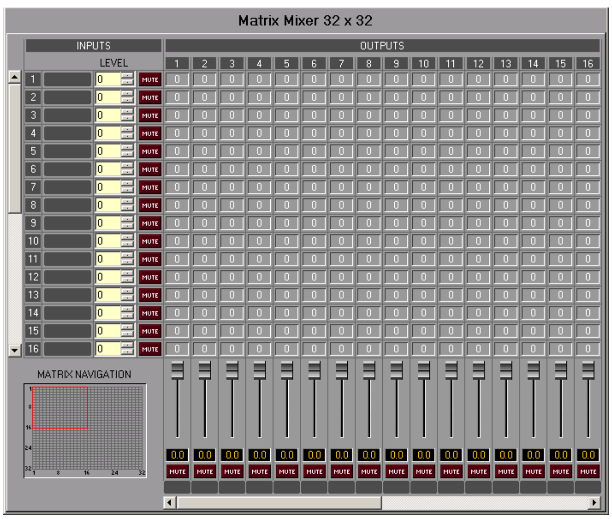
Element	Default	Range	Description
			Switches between frequency (magnitude) and phase response (phase) indication.
			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
	6dB	6dB, 12dB	SLOPE sets the transition band slope of the low shelf filter.
	1000Hz	20 Hz...20 kHz	FREQUENCY sets the cut-off frequency of the low shelf filter.
	0.0 dB	-18...+18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of the low shelf filter.
			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the low shelf filter has on the sound.

Filter Editing via "Mouse Movement" in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the filter's frequency by moving the mouse to the left or to the right.

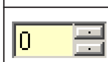
MATRIX MIXER



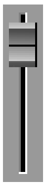
The DSP block Matrix Mixer allows connecting inputs and outputs. Left clicking the node in the matrix where the output channel's column and the input channel's line meet with the mouse does connect an output to an input. Again clicking onto the corresponding node disconnects inputs and outputs. Right clicking onto a node opens a dialog box for setting levels.
Making connections is not restricted in any way, e.g. connecting various inputs to a single output is as possible as connecting a single input to a variety of outputs. The notation I x O represents a matrix with I inputs and O outputs.



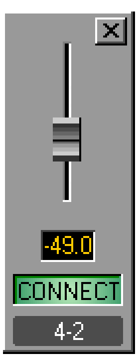
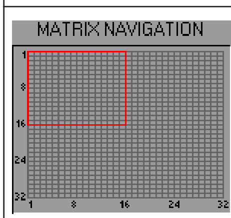
Inputs

Element	Default	Range	Description
			Fixed channel labeling: channels of a matrix of the dimension I x O are numbered from 1 to I.
			Text field for providing an input channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**
	0 dB	-80...0 dB	Setting an input channel's signal level. The desired value can be entered in dB. **HINT: Ramping is deactivated by default.**
			MUTE button mutes the input channel

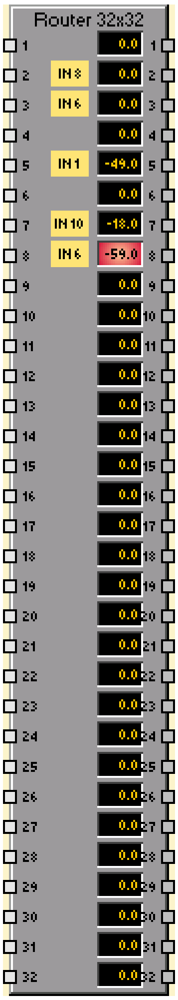
Outputs

Element	Default	Range	Description
	0 dB	-80...0 dB	Fader for setting the corresponding output channel’s level.
	0 dB	-80...0 dB	Fader Display. This field indicates the current fader setting as a numerical value. Entering the desired value in dB is possible as well.
			MUTE button mutes the output signal.
			Text field for providing an output channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

Nodes

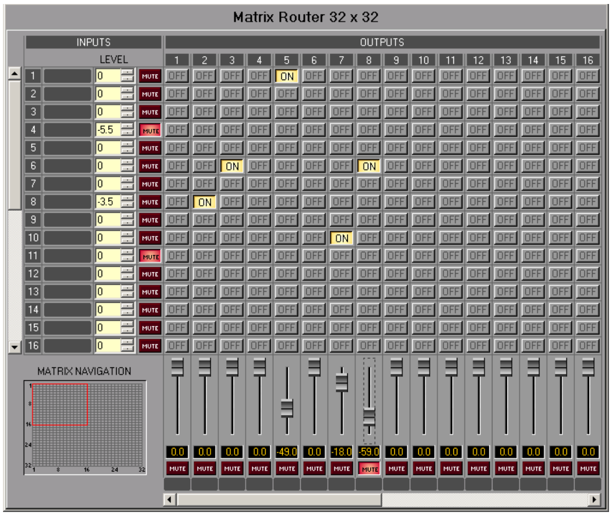
Element	Description
	Right clicking a node opens this dialog with a fader and fader display for setting the level and a CONNECT-Button for setting or resetting the node’s connection. The label represents the node’s position ("line - column") within the matrix.
	This area is navigating through largely dimensioned matrixes. The red rectangle marks the currently indicated matrix area. Changing the displayed area is possible by dragging the rectangle with the left mouse button pressed down. MATRIX NAVIGATION appears only when the complete matrix view does not fit in the screen.

MATRIX ROUTER



The DSP block Matrix Router allows connecting an output to only one input. Left clicking the node in the matrix where the output channel's column and the input channel's line meet with the mouse does connect an output to an input. Again clicking onto the corresponding node disconnects inputs and outputs.

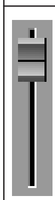
Matrix Routers are subject to a restriction. Only one input can be connected to an output at a time. However, connecting an input to various outputs is possible. A Matrix Router can distribute the signal of an input. Mixing input signals is not possible. The notation I x O represents a matrix with I inputs and O outputs. The number of inputs (2 to 32) and outputs (2 to 32) can be selected when adding the DSP block to the configuration.

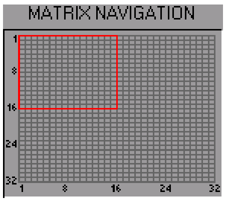


Inputs

Element	Default	Range	Description
			Fixed channel labeling: channels of a matrix with the dimensions I x O are numbered from 1 to I.
			Text field for providing an input channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**
	0 dB	-80...0 dB	Setting the signal level of an input channel. Entering the desired value in dB is possible. **HINT: Ramping is deactivated by default.**
			MUTE button mutes the input channel.

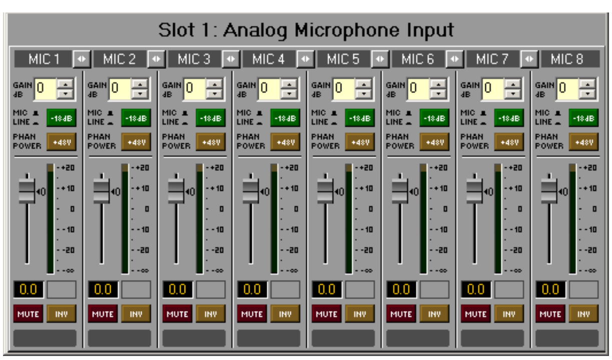
Outputs

Element	Default	Range	Description
	0 dB	-80...0 dB	Fader for setting the corresponding output channel’s level.
	0 dB	-80...0 dB	Fader Display. This field indicates the current fader setting as a numerical value. Entering the desired value in dB is possible as well.
			MUTE button mutes the output signal.
			Text field for providing an output channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

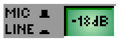
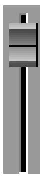
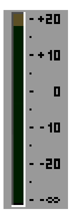
Element	Description
	This area is navigating through largely dimensioned matrixes. The red rectangle marks the currently indicated matrix area. Changing the dis- played area is possible by dragging the rectangle with the left mouse button pressed down. MATRIX NAVIGATION appears only when the complete matrix view does not fit in the screen.

ANALOG MICROPHONE INPUT

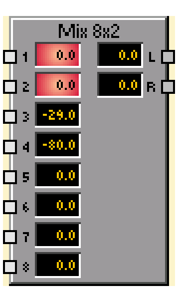
The DSP block Analog Microphone Inputs provides access to the eight analog inputs with microphone in- put sensitivity of a MI-1 Microphone Input card. Establishing independent settings for each input channel is possible. Equivalent to the ones of the Analog Line Inputs DSP block, these settings include a fader with ramping function as well as Mute and Invert buttons. In addition, setting the GAIN in steps of 6 dB is possible in a range between 0 dB and +60 dB. +48 V phantom power can be activated per channel. Adjacent channels can be linked by use of the LINK button. This allows convenient synchronous settings of several input channels, e.g. for stereo signals. Up to three MI-1 can be used simultaneously in a N8000 or P 64.



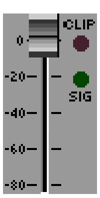
Element	Default	Range	Description
			Permanent channel labeling. Channels are numbered from MIC 1 to MIC 8.
			LINK button for linking (grouping) adjacent input channels. The Link button does not affect Gain, PAD and Phantom Power settings. Changing any of these three parameters in a linked channel does not automatically change the parameters' setting in any of the other grouped channels.
	0.0 dB	0...60 dB	The input channel gain can be adjusted in steps of 6 dB.

			The -18dB button (PAD) is for switching between microphone and line input sensitivity.
			The +48V button is for activating phantom power whenever a suitable condenser microphone is being used.
	0.0 dB	-80...+18.0 dB	Fader for setting the input level.
	0.0 dB	-80...+18.0 dB	The fader display shows the numerical value of the current fader setting and additionally allows entering a desired value.
			Indicates the input signal’s current level.
			MUTE button for muting the input signal.
			INV button for inverting (180° phase shift) the input signal’s polarity.
			Text field for labeling an input channel with an exclusive IRIS-Net name. **CAUTION: Using * (asterisk) or = (equal) signs in a name is not permissible.**

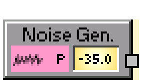
MIXER



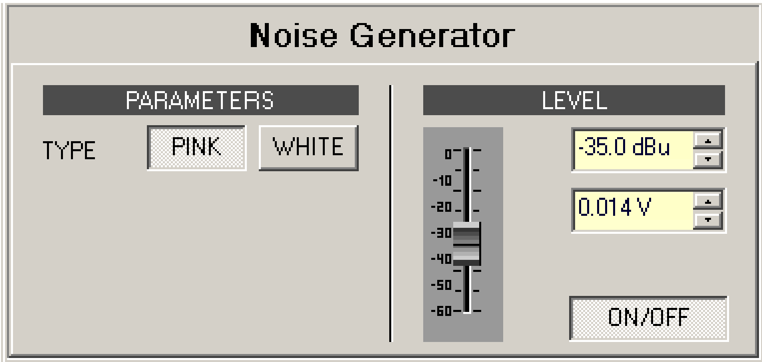
The DSP block Mixer allows mixing various input signals to output them via a single output (mono) or two outputs (stereo). Monaural and stereo mixers with 2, 4, 8, 16, 24 and 32 inputs are available.

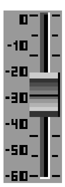
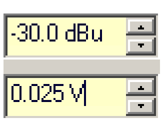
Element	Default	Range	Description
			Fixed channel naming: input channels are numbered from CH.1 to CH.n. Output channel(s) are labeled OUT (Mono-Mixer) or L and R (Stereo-Mixer).
			LINK button for connecting (grouping) neighboring input channels.
			Stereo Balance (Stereo-Mixer only) defines the percentage of the input signal being distributed to the left or the right.
			INV button for inverting the input channel.
			SOLO button for monitoring a single input signal.
			MUTE button for muting the corresponding input signal.
	0.0 dB	-80... 0 dB	Fader for setting the levels in corresponding inputs or outputs. If a signal is present at the channel (above -40dBu), the "SIG" LED lights. The "CLIP" LED additionally lights when the signal level nears clipping (+21dBu).
			The Fader Display indicates the current fader setting as a numerical value and allows entering the desired value.
			Text field for labeling an input channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

NOISE GENERATOR

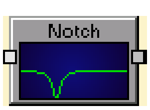


The DSP block Noise Generator generates white or pink noise. Pink noise has a spectral dispersion with constant power per relative bandwidth, whereas the octave from 20 to 40 Hz has the identical noise power as the octave between 10000 and 20000 Hz. Every time the frequency is doubled the power is cut in half.
White noise has a spectral dispersion with constant power per absolute bandwidth, stated in Hz. The 20 Hz range between 20 and 40 Hz has the same noise power as the 20 Hz range between 10000 Hz and 10020 Hz.

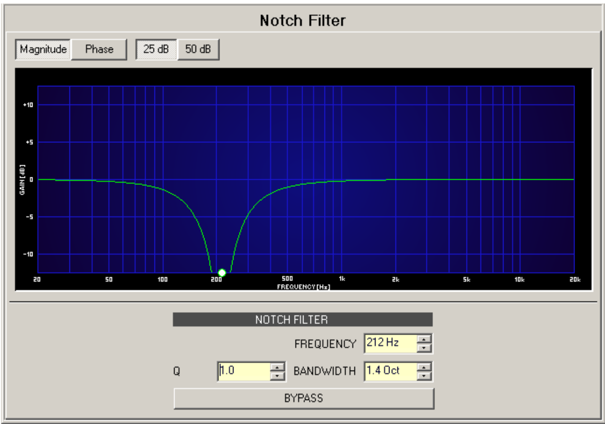


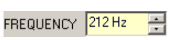
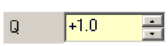
Element	Default	Range	Description
			The PINK button selects "Pink Noise" mode.
			The WHITE button selects "White Noise" mode.
	-35 dB	-60... 0 dB	Fader for setting the noise signal’s level.
	-35.0 dBu or 0.014 V	-60...0 dBu or 0.001...0.775 V	Entering the signal level is possible in dBu or V.
			Switch for activating or deactivating the noise generator.

NOTCH FILTER



Notch filters pass all frequencies except for the notch frequency, which they stop completely. Since it is not realistically possible to create a perfect filter that stops one frequency completely and passes all other frequencies totally unaltered, notch filter design involves compromises that allow some (adjustable)width in the notch and less than infinite attenuation at the notch frequency.

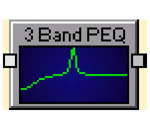


Element	Default	Range	Description
			Switches between frequency (magnitude) and phase response (phase) indication.
			Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
	1000Hz	20 Hz...20 kHz	FREQUENCY sets the center frequency (notch frequency) of the notch filter.
	1	0.4...100	The Q-value sets the quality and thus the response of the notch filter.
	1.4	0.1...3	BANDWITH sets the quality and thus the response of the notch filter.
			BYPASS switches the filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that the notch filter has on the sound..

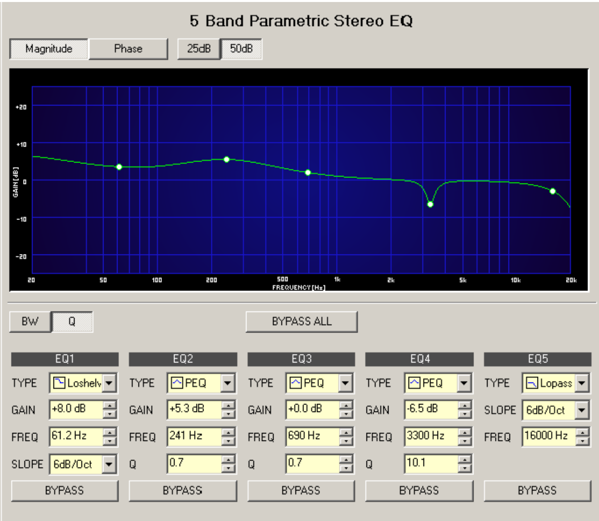
Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the filter’s frequency by moving the mouse to the left or to the right. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of the filter.

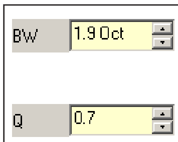
PARAMETRIC EQ



Equalizers accentuate or lower the audio signal within specific frequency ranges. IRIS-Net provides parametric 3-, 5-, 7- as well as 12-Band equalizers in monaural (1 input and 1 output) and stereo quality (2 inputs and 2 outputs) plus, in addition, PEQs with a freely selectable amount of filters between 1 and 32.



Element	Default	Range	Description
Magnitude Phase			Switch for displaying amplitude frequency response (magnitude) or phase response (phase)
25dB 50dB			Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
BW Q			Switch for changing between bandwidth BW and Quality Q when setting employed filters.
BYPASS ALL			Pressing BYPASS ALL switches of all filters.
EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. His- helv, Hipass, Lopass, Allpass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope. – Allpass is a filter that has no influence on the frequency response but on the phase response in the transmission function.
GAIN +0.0 dB	0 dB	-18...+18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
FREQ 30.0 Hz	depends on number of filters	20 Hz...20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.

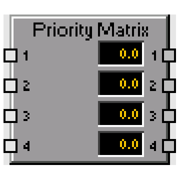
	1.9 Oct or 0.7	0.04...2.8 6 Oct. or 0.4...40	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct.
	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6- alignment, which describes a drastic rise in the cut-off frequency range.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its amplification (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

PRIORITY MATRIX



The Priority Matrix DSP block is an extended Matrix Mixer. Clicking with the left mouse button onto the node in the matrix where the output channel’s column and the input channel’s line meet, connects an output to an input. Again clicking onto the node separates the connection. Right clicking on a node allows editing the level. When connecting the device to a PROMATRIX/PROANNOUNCE System, allows dynamically setting nodes in the Priority Matrix from call stations. Please also refer to the notes in chapters RS-232 Setup and Macros.

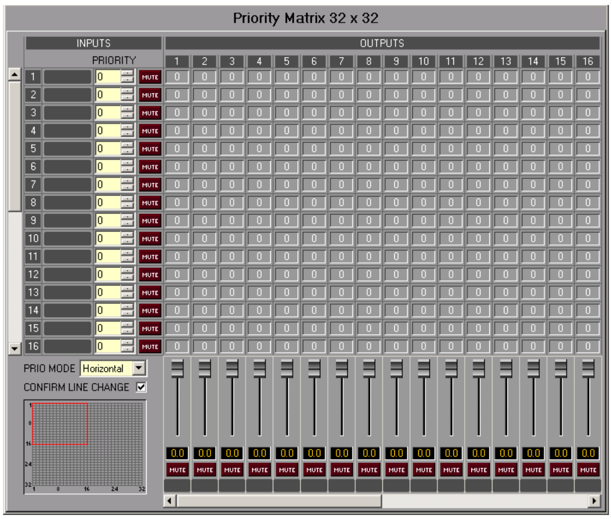
Next to connecting inputs and outputs, assigning Priorities to input channels is possible as well. The Priority Matrix provides the following features:

- Exactly 256 Priorities exist, numbered from 0 up to 255.
- 1 represents the lowest priority and 255 is the highest priority.
- If various input signals with different priorities are assigned to a single output at the same time, only the signal with the highest priority is connected through to the output.
- If input signals with equal priority are assigned to a single output at the same time, a mixed signal of these input signals is output.
- An input signal with priority 0 assigned to an output gets connected through to the output in any case. If another signal (with another priority) is present at the output, it is mixed with the signal that has priority 0.

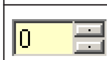
For priority mode "horizontal" only:

– If input signals of different priority (>0) are assigned to different outputs, only the input signal with the highest priority is connected through to the selected output(s). No signal is present at outputs that are only connected to inputs with low priority, this means that set nodes of inputs with low priority are ignored in priority mode "horizontal".

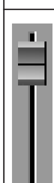
Setting connections is not restricted in any way, e.g. connecting various inputs to a single output as well as assigning a single input to a variety of outputs is possible. The notation I x O represents a matrix with I inputs and O outputs.



Inputs

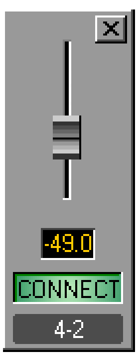
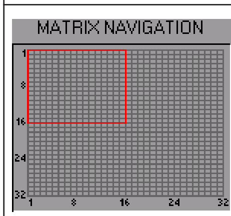
Element	Default	Range	Description
			Fixed channel labeling: channels are numbered from 1 to I.
			Text field for providing an input channel with an internal IRIS-Net name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**
	0	0...255	The Priority field shows the input channel's set priority. Entering the desired priority in the range between 0 and 255 is possible.
			MUTE button mutes the input signal.

Outputs

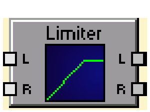
Element	Default	Range	Description
	0 dB	-80...0 dB	Fader for setting the corresponding channel's output level.

	0 dB	-80...0 dB	Fader-Display. This field indicates the current fader setting as a numerical value. Entering the desired value in dB is possible as well.
			MUTE button mutes the output signal.
			Textfeld zur Beschriftung des Eingangskanals mit einer IRIS-Net-internen Bezeichnung. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**

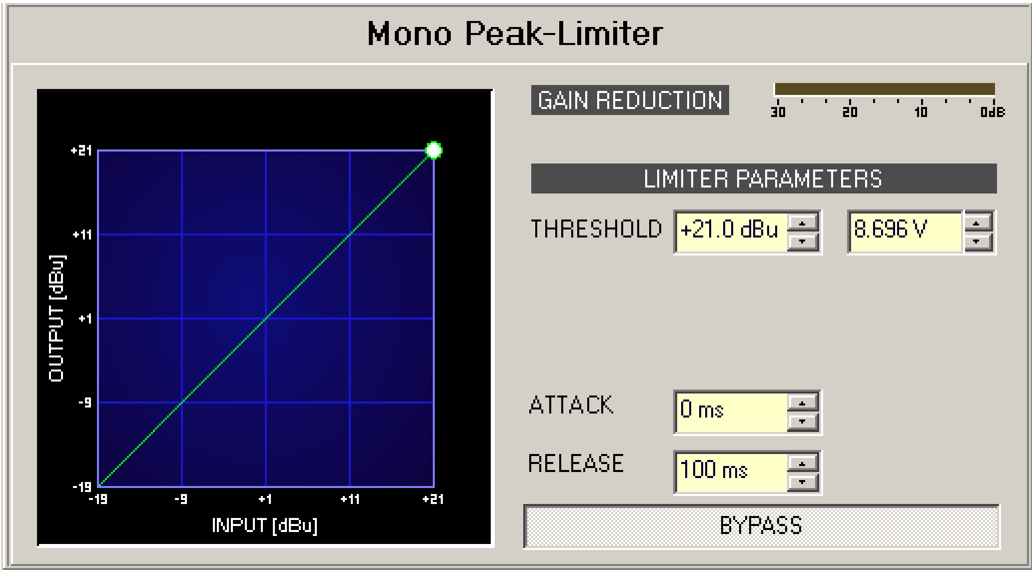
Nodes

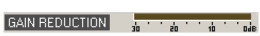
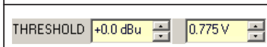
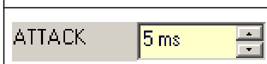
Element	Description
	Selecting priority mode of matrix. Available modes are "horizontal" and "vertical".
	If the checkbox is selected a confirmation prompt appears every time nodes are set or reset.
	Right clicking a node opens this dialog with a fader and fader display for setting the level and a CONNECT-Button for setting or resetting the node's connection. The label represents the node's position ("line - column") within the matrix.
	This area is navigating through largely dimensioned matrixes. The red rectangle marks the currently indicated matrix area. Changing the dis- played area is possible by dragging the rectangle with the left mouse button pressed down. MATRIX NAVIGATION appears only when the complete matrix view does not fit in the screen.

PEAK-LIMITER

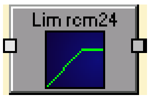


A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage.

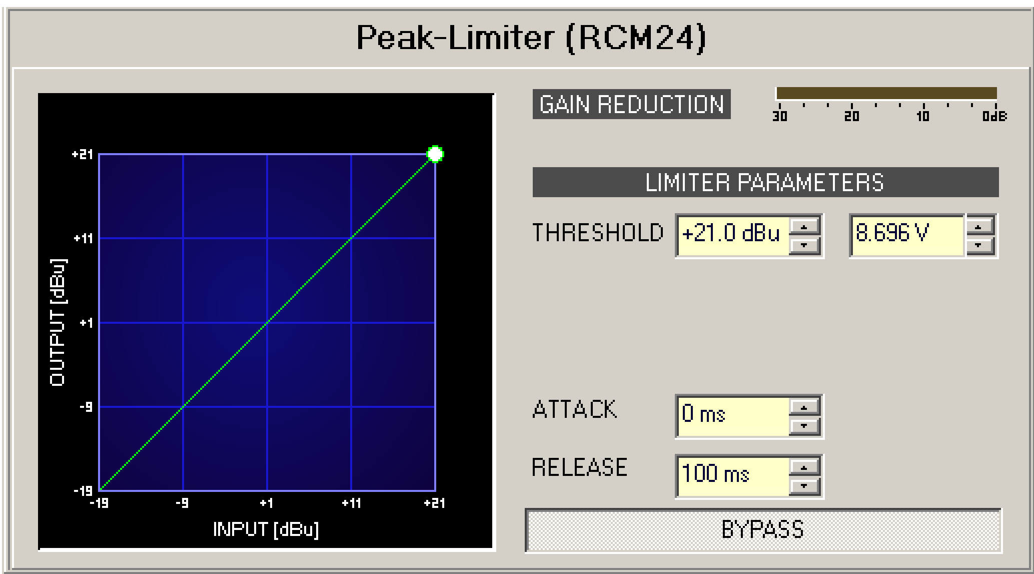


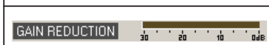
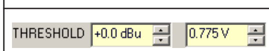
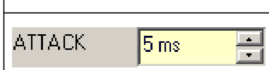
Element	Default	Range	Description
			This indicator signals in dB by how much the limiter reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.
	0.0 dBu or 0.775 V	-9.0...+21.0 dBu or 0.275...8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter unaffected. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Generally, the THRESHOLD of the limiter should be set some dB higher than the compressor's threshold to effectively limit high level peaks. Entering the threshold value is possible in dBu or V. The value can be entered in either box and will automatically be converted in the other.
	5 ms	0...50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.
	250 ms	10...1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

PEAK-LIMITER (RCM-24)



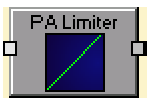
A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them in an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage. The characteristics of this Limiter are identical to those of the limiters used in Precision Series power amps.



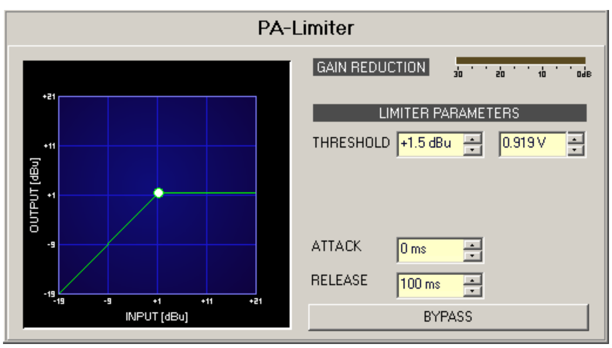
Element	Default	Range	Description
			This indicator signals in dB by how much the limiter reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.
	0.0 dBu or 0.775 V	-9.0...+21.0 dBu or 0.275...8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter unaffected. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Generally, the THRESHOLD of the limiter should be set some dB higher than the compressor’s threshold to effectively limit high level peaks. Entering the threshold value is possible in dBu or V. The value can be entered in either fox and will automatically be converted in the other.
	5 ms	0...50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.

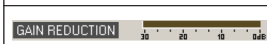
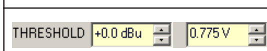
	250 ms	10...1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

PEAK ANTICIPATION (PA) LIMITER



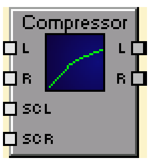
A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them in an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage. The Peak Anticipation Limiter calculates the gain reduction using the maximum value of the look ahead buffer and is optimized for loudspeaker and transducer protection.



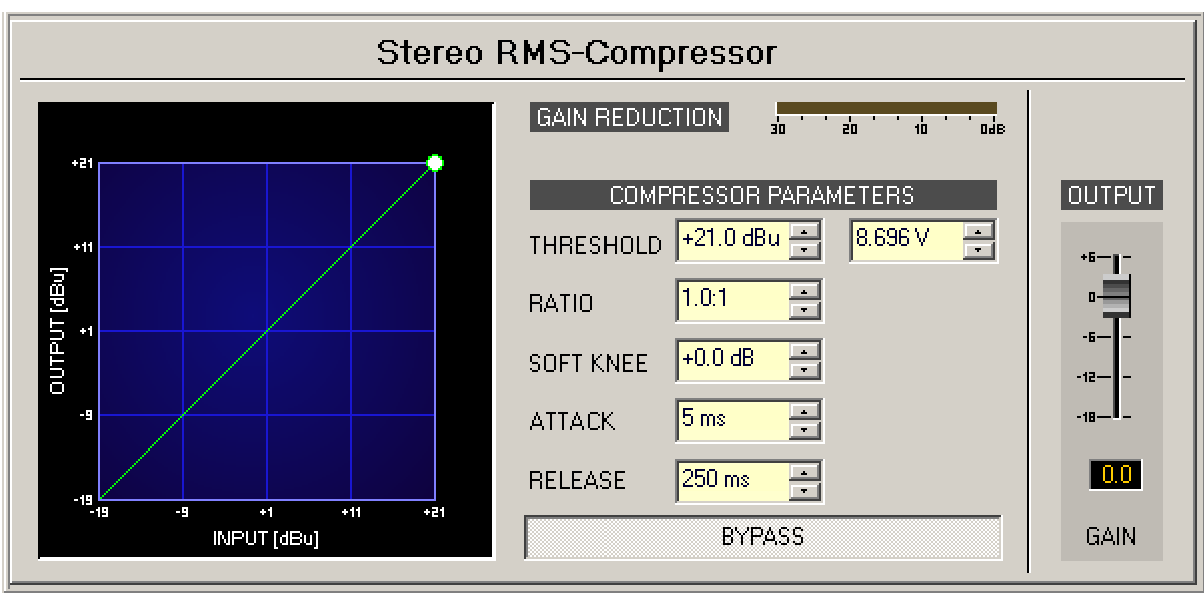
Element	Default	Range	Description
			This indicator signals in dB by how much the limiter reduces the signal level. A red bar that increases from the right to the left indicates the degree of gain reduction.
	21 dBu or 8.696 V	-9.0...+21.0 dBu or 0.275...8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Generally, the THRES- HOLD of the limiter should be set some dB higher than the compressor's threshold to effectively limit high level peaks. Entering the threshold value is possible in dBu or V. The value can be entered in either box and will automatically be converted in the other.
	0 ms	0...50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.

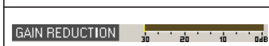
	100 ms	10...1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

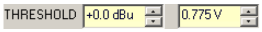
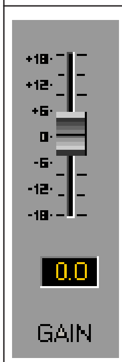
RMS-COMPRESSOR



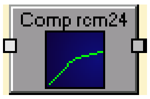
The compressor reduces the dynamic range of audio signals. Once the signal exceeds a certain threshold, the signal gets compressed, i.e. major input level changes result in minor output level changes. Narrowing the dynamic range often allows for easier recording or mixing the audio signal.
Compressors with effective value analysis in monaural (1 input and 1 output) and stereo quality (2 inputs and 2 outputs) are provided. A side channel input allows feeding the compressor with an external control signal, which, for example, can also be the input signal processed by an equalizer. As a result the compressor reacts to specific frequency bands more than to others. However, using a totally different trigger signal lets you achieve various special effects.



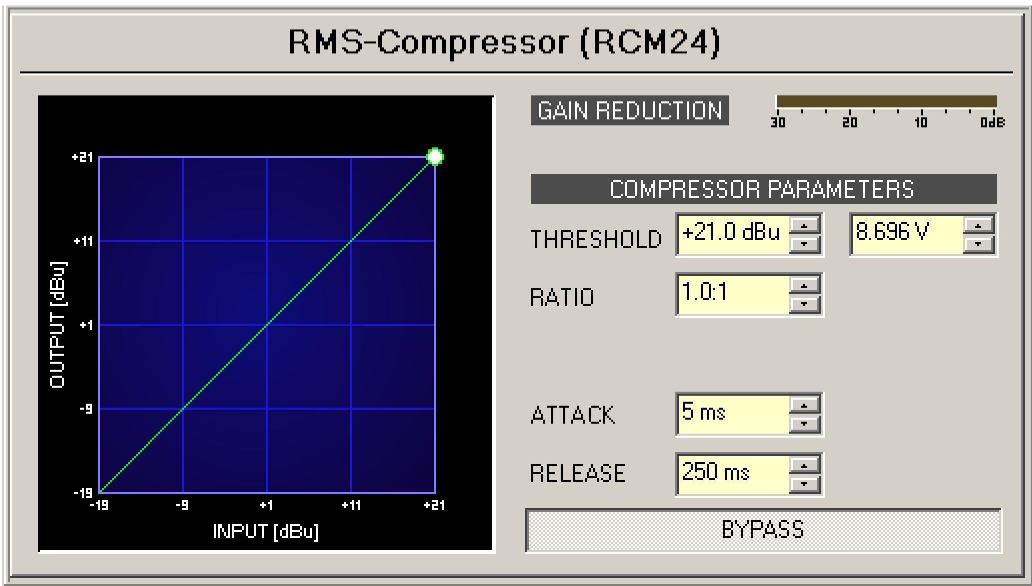
Element	Default	Range	Description
			The attenuation (gain reduction) indicator signals the degree of level reduction in dB. A red bar that increases from right to left indicates the degree of gain reduction.

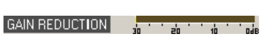
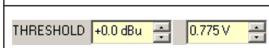
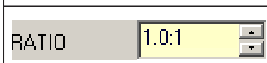
	0.0 dBu or 0.775 V	-9.0...+21.0 dBu or 0.275...8.696 V	THRESHOLD defines the signal level at which the Compressor sets in. Entering the desired value is possible in dBu as well as in V. The entered value is automatically converted in both directions.
	1.0:1	1.0:1...100.0:1	RATIO defines the compression rate, i.e. the degree of compression above the threshold level. For example, a rate of 4.0 : 1 represents a signal reduction by factor 4.
	0.0 dB	0.0...20 dB	SOFT KNEE has an influence on the curve's bend, making it possible to avoid a sudden start of compression once the signal exceeds the THRESHOLD. When set- ting the parameter to a value of n dB, slight compression already sets in at an input signal level of THRESHOLD - n/2. If the signal level exceeds THRESHOLD + n/2, the signal is compressed with the set RATIO. The compression rate increases slowly until RATIO is reached in the range between these two values.
	5 ms	5...150 ms	ATTACK defines the velocity, at which the compressor sets in. A short attack rate means that even short signal peaks are efficiently compressed. Longer attack rate leave signal peaks untouched.
	250 ms	20...2000 ms	RELEASE defines the control time interval the compressor takes to return to an uncompressed signal level, after the signal dropped below the set threshold.
			BYPASS activates (not engaged) or deactivates the Compressor (engaged), which allows for quick A / B comparison between the compressed and uncompressed audio signal.
	0.0 dB	-18.0...+6.0 dB	GAIN defines the compressor's output amplification. GAIN allows matching input and output levels if compression lead to a level reduction of the output signal.

RMS-COMPRESSOR (RCM-24)



The characteristics of the Compressor (RCM-24) offered by IRIS-Net is identical to the compressors used in Precision Series power amps.

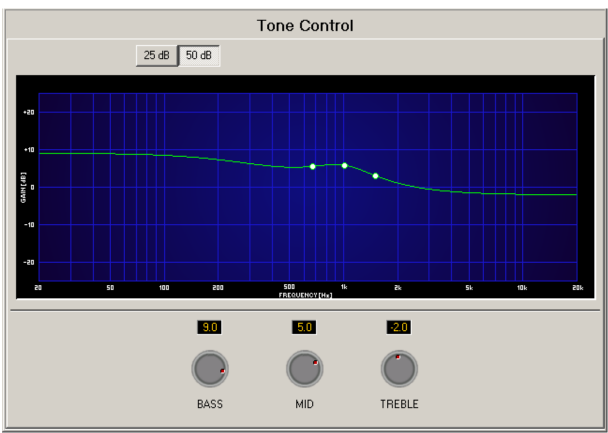


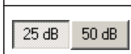
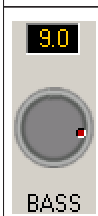
Element	Default	Range	Description
			This bar meter indicates gain reduction in dB, i.e. by how much the Compressor reduces the signal level. A red bar that increases from right to left indicates the degree of gain reduction.
	0.0 dBu or 0.775 V	-9.0...+21.0 dBu or 0.275...8.696 V	THRESHOLD defines the signal level at which the Compressor sets in. Entering the desired value is possible in dBu as well as in V. The entered value is automatically converted in both directions.
	1.0:1	1.0:1, 1.4:1, 2.0:1, 4.0:1, 8.0:1	RATIO defines the compression rate, i.e. the degree of compression above the threshold level. For example, a rate of 4.0 : 1 represents a signal reduction by factor 4.
	5 ms	5...150 ms	ATTACK defines the velocity, at which the compressor sets in. A short attack rate means that even short signal peaks are efficiently compressed. Longer attack rate leave signal peaks untouched.

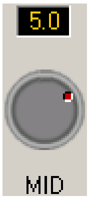
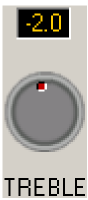
	250 ms	20...2000 ms	RELEASE defines the control time interval the compressor takes to return to an uncompressed signal level, after the signal dropped below the set threshold.
			BYPASS activates (not engaged) or deactivates the Compressor (engaged), which allows for quick A / B comparison between the compressed and uncompressed audio signal.

TONE CONTROL

The DSP block Tone Control has three filters with rotary controls to easily control the tone of the audio signal. Each control has affect on the gain of an otherwise pre-defined filter. Making complex filter settings is not necessary. Operation is straightforward and known from the 3-band tone controls in a mixer's input channels.

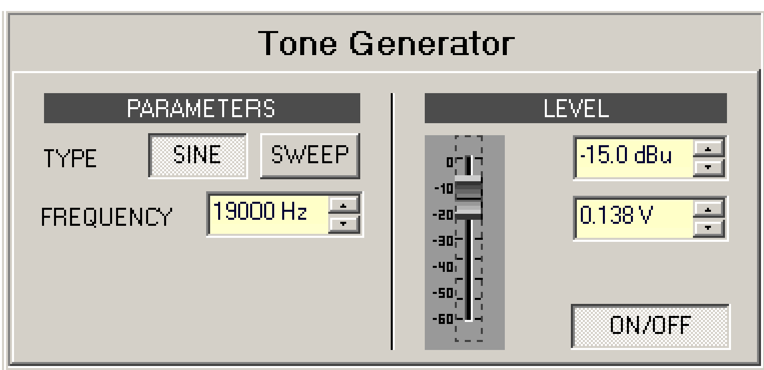


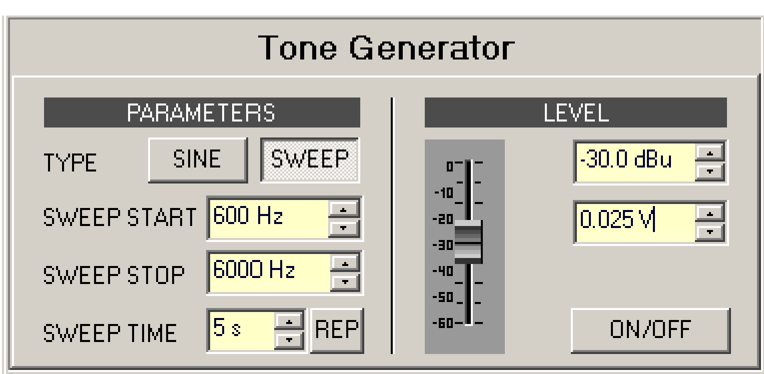
Element	Default	Range	Description
			Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or 50 dB (± 25 dB).
	0 dB	-12...+12 dB	BASS changes (amplifies or attenuates) the gain of a Bass filter (Low-Shelving filter). The filter's cut-off frequency is fixed to 666,6 Hz. The desired amplification or attenuation can be set by directly entering the desired value or by using the rotary control.

	0 dB	-12...+12 dB	MID changes (amplifies or attenuates) the gain of a Mid-EQs (Peak-Dip-Filter). The filter's center frequency is fixed to 1 kHz. The desired amplification or attenuation can be set by directly entering the desired value or by using the rotary control.
	0 dB	-12...+12 dB	TREBLE changes (amplifies or attenuates) the gain of a Treble filter (High-Shelving-Filter). The filter's cut-off frequency is fixed to 1,5 kHz. The desired amplification or attenuation can be set by directly entering the desired value or by using the rotary control.

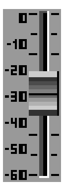
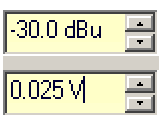
TONE GENERATOR

Operating the DSP block Tone Generator is possible in two different ways. Sine mode generates a sine signal with a constant frequency. Sweep mode generates a one-time or periodical sine sweep over a specific frequency range.





Element	Default	Range	Description
			The SINE button puts the tone generator in constant sine signal mode.

			The SWEEP button puts the tone generator in sine sweep mode.
			The constant frequency of a sine signal.
			The start frequency of a sine sweep.
			The stop frequency of a sine sweep.
			The time in seconds, it takes the sine signal to sweep through the range marked by the start frequency and the stop frequency.
			Activates the periodical repetition of a sine sweep.
	-35 dB	-60... 0 dB	Fader for setting the signal level.
	-35.0 dBu or 0.014 V	-60...0 dBu or 0.001...0.775 V	Entering the signal level is possible in dBu or V.
			Switch for activating or deactivating the tone generator.

ASCII Control Protocol

RS-232 SETTINGS

The RS-232 interface of the device is preconfigured for full duplex operation. Set values are:

Parameter	**Setting**
Baud Rate	19200 bits per second
Data Bits	8
Parity	None
Stop Bits	1
Flow Control	Xon / Xoff

The command string "*** <device> command mode entered ***" is sent to RS-232 once the device is powered up and has completed its boot sequence. The RS-232 interface is now ready for communication.

ETHERNET SETTINGS

Factory set values of the Ethernet port are:

Parameter	**Setting**
IP address	192.168.1.100
Network mask	255.255.255.0

Standard gateway	192.168.1.1
Telnet port	23
Username	netmax
Password	netmax

A Telnet session can be established with an application such as Windows HyperTerminal. Log in requires the username and password (default values given in above table) after which the ASCII Control Protocol can be started with the command "parser". If there is already another ASCII Control Protocol up and running, a new ASCII Control Protocol session can be forced by using the command "parser -f". The command string "*** <device> command mode entered ***" will be displayed when the ASCII Control Protocol has started successfully. The device is now ready for ASCII Control Protocol communication.

ASCII CONTROL PROTOCOL

A simple ASCII string protocol, which is referred to as ASCII Control Protocol is implemented in the device. Commands are organized in a tree structure with up to 5 levels. The slash "/" or a space " " can be used for separation. The question mark "?" can be utilized to query parameter settings or commands of the corresponding level. To step down one level you have to enter "..". Use "/" to get back to level 1.
The following table lists the ASCII Control Protocol commands with brief explanations.

Commands for RS-232 communication

			Read Write	**Values**	**Description**
/COMM	/LINEFEED		R / W	ON, OFF	Linefeed state for RS-232 communication
	/PROMPT		R / W	ON, OFF	Prompt state for RS-232 communication
	/ECHO		R / W	ON, OFF	Echo state for RS-232 communication
	/LOG	/state	R / W	ALL_ON, COMMANDS_ON, OFF	Log state for RS-232 communication
		/CLEAR	execute	none	Deletes the existing log file
		/PRINT	execute	none	Displays the existing log file
		/SAVE	execute	none	Saves the log to a file "parser.log" in directory /flash/log

Commands for CAN-BUS communication

				Read Write	Values	Description
/PARAM	/CAN	/MONITOR	/VALUE	R	error! (unknown channel), INPUT_A RCM: X, INPUT_B RCM: X,OUTPUT_A RCM:X, OUTPUT_B RCM: X	Input or output channel of amplifier (with CAN address X) currently selected for monitor bus
		/SCANSTATE	/VALUE	R	SCAN_FAILED, SCAN_READY	State of CAN-BUS after scan
		/FUNCTIONS	/CAN_RESET	exec	none	Reset of CAN-BUS
			/CAN_SCAN	exec	none	Scan CAN-BUS for new devices
			/SET_BAUD_ ALL	exec	10, 20, 63, 125, 250, 500	Set new Baud rate of the device and all devices connected to the CAN-BUS
			/SET_ MONITOR	exec	input_a, input_b, output_a, output_b; 1-250	Selects Monitor Bus of a remote amplifier

Commands for system queries

				Read Write	Values	Description
/PARAM	/LOCAL	/ CHASSISTEMP	/VALUE	R		Device temperature in degree Celsius
		/FANSPEED	/VALUE	R / W	0, 1, 2, 3	Fan speed (0 = off, 1 = slow, 2 = mid, 3= fast)
		/GPIANALOG	/ IDX1...IDX4	R	0, 1	State of GPI using analog circuit
		/GPIDIGITAL	/ IDX1...IDX4	R	0, 1	State of GPI using digital circuit
		/GPO	/ IDX1...IDX3	R / W	0, 1	State of GPO (0 = open, 1 = closed)
		/ REMOTEFAUL T	/ IDX1...IDX1 00	R	0, 1	State of remote device (0 = OK, 1 = does not answer)
		/ CONDRESULT	/ IDX1...IDX1 00	R	0, 1	State of condition (0 = false, 1 = true)

		/REMOTEMASTERFAULT	/IDX1...IDX100	R	0, 1	State of Masterfault of remote device (0 = not set, 1 = set)
		/USERFAULT	/IDX1...IDX8	R / W	0, 1	Status of user faults

Commands for CobraNet

				Read Write	Values	Description
/PARAM	/COBRANET	/CONDUCTORSTATUS	/VALUE	R	0, 1	Shows if the device is Conductor in CobraNet
		/CONDUCTORPRIORITY	/VALUE	R / W	0...255	Conductor priority in CobraNet
		/RXBUNDLE	/IDX1...IDX4	R / W	0...65535	Number of received Bundle
		/RXBUNDLENAME	/IDX1...IDX4	R / W	max. 16 characters	Name of received Bundle
		/RXCHANNELNAME	/IDX1...IDX32	R / W	max. 16 characters	Name of channels of received Bundles
		/RXCLIP	/IDX1...IDX32	R	0, 1	Clipping LED of channels of received Bundles
		/RXMUTE	/IDX1...IDX32	R / W	0, 1	Mute of channels of received Bundles
		/RXSIGNAL	/IDX1...IDX32	R	0, 1	Signal LED of channels of received Bundles
		/RXSTATUS	/IDX1...IDX4	R	0, 1	Status of received Bundle (0 = NOT RECEIVED, 1 = ACTIVE)
		/SYSNAME	/VALUE	R / W	max. 60 characters	Name of device in CobraNet
		/TXBUNDLE	/IDX1...IDX4	R / W	0...65535	Number of sent Bundle

		/ TXBUNDLENAME	/ IDX1...IDX4	R / W	max. 16 characters	Name of sent Bundle
		/ TXCHANNELNAME	/ IDX1...IDX32	R / W	max. 16 characters	Name of channels of sent Bundles
		/TXCLIP	/ IDX1...IDX32	R	0, 1	Clipping LED of channels of sent Bundles
		/TXMUTE	/ IDX1...IDX32	R / W	0, 1	Mute of channels of sent Bundles
		/TXSIGNAL	/ IDX1...IDX32	R	0, 1	Signal LED of channels of sent Bundles

Commands for AI-1 Module

					Read Write	Values	Description
/ PARAM	/DSP	/ ANALOGIN_x	/ CHANNELINFO	/VALUE	R / W	0...127	Linking of inputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-Buttons.
			/GAIN	/ IDX1...IDX8, ALL	R / W	-80...20	Gain of input
			/MUTE	/ IDX1...IDX8, ALL	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/ IDX1...IDX8, ALL	R / W	0, 1	0 = normal, 1 = inverted
			/NAME	/ IDX1...IDX8, ALL	R / W	max. 16 characters	Input name
			/RAMPTIME	/ IDX1...IDX8, ALL	R / W	0.001...20	Ramping Time of input (seconds)

			/ PILOTDETECT .FREQ	/ IDX1...IDX 8, ALL	R / W	20...2 0000	Frequency of pilot tone to detect
			/ PILOTDETECT .FLAG	/VALUE	R	0x00... 0xFF	Status of pilot tone detection of all inputs (1 = pilot tone detected, 0 = pilot tone not detected). The last of the four hexadecimal values corresponds in binary representation to the 8 inputs.
			/ PILOTDETECT .ENABLE	/VALUE	R	0x00... 0xFF	Activation of pilot tone detection of all inputs. The last of the four hexadecimal values corresponds in binary representation to the 8 inputs.

Commands for MI-1 Module

					Read Write	Values	Description
/ PARA M	/DS P	/ ANALOGMI CIN_x	/GAIN	/ IDX1...IDX 8, ALL	R / W	-80...18	Gain of input
			/MUTE	/ IDX1...IDX 8, ALL	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/ IDX1...IDX 8, ALL	R / W	0, 1	0 = normal, 1 = inverted
			/ CHANNELIN FO	/VALUE	R / W		Linking of inputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-Buttons.
			/NAME	/ IDX1...IDX 8, ALL	R / W	max. 16 character s	Input name
			/RAMPTIME	/ IDX1...IDX 8, ALL	R / W	0.001... 20	Ramping Time of input (seconds)
			/MIC.LINE	/ IDX1...IDX 8, ALL	R / W	0, 1	Selects input level (0 = Mic level, 1 = Line level)

			/MIC.PHANPOWER	/IDX1...IDX8, ALL	R / W	0, 1	Phantom Power (+48 V) of input (0 = off, 1 = on)
			/MIC.GAIN	/IDX1...IDX8, ALL	R / W	0...10	Microphone gain in steps of 6dB.

Commands for AO-1 Module

					Read Write	Values	Description
/PARAM	/DSP	/ANALOGOUT_x	/GAIN	/IDX1...IDX8, ALL	R / W	-80...18	Gain of outputs
			/MUTE	/IDX1...IDX8, ALL	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/IDX1...IDX8, ALL	R / W	0, 1	0 = normal, 1 = inverted
			/NAME	/IDX1...IDX8, ALL	R / W	max. 16 characters	Output name
			/RAMPTIME	/IDX1...IDX8, ALL	R / W	0.001...20	Ramping Time of output (seconds)

Commands for Automatic Gain Control (AGC)

					Read Write	Values	Description
/PARAM	/DSP	/AGC_x	/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/DECREASE_TIME	/VALUE	R / W	300...20000	GAIN DECREASE time in ms
			/GAIN	/VALUE	R / W	-6...18	Target Level in dB
			/GAINREDUCTION	/VALUE	R	-30...30	Current Gain change as full scale value (32 bit) Transform given value into dB by: 20 log (0x7FFFFF/ value)
			/HOLD	/VALUE	R / W	50...60000	Hold time in ms

			/ INCREASE_ TIME	/VALUE	R / W	30...20000	GAIN INCREASE time in ms
			/KNEE	/VALUE	R / W	0.1...15	Knee in dB
			/RATIO	/VALUE	R / W	1...15	Compressor ratio
			/RELEASE	/VALUE	R / W	300...20000	Release time in ms
			/ THRESHOL D	/VALUE	R / W	-30...0	Threshold in dB (example: -8.5)

Commands for Ambient Noise Control (ANC)

						Read Write	Values	Description
/ PARA M	/DS P	/ ANC_x	/ AMBABOV ETH- RESH	/VALUE		R	0, 1	0 = Ambient level below Threshold, 1 = Ambient level above Threshold
			/BYPASS	/VALUE		R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/ DECREAS ETIME	/VALUE		R / W	10...3000	Time for GAIN DECREASE in ms
			/ FREEZEG AIN	/VALUE		R / W	0, 1	Freeze current gain values (0 = variable, 1 = freeze)
			/GAIN	/ IDX1...I DXn	/ VALU E	R / W	-60...0	Gain of inputs
			/ GAINMAX	/VALUE		R / W	-29.9...30	Maximum gain of output signals
			/GAINMIN	/VALUE		R / W	-30...29.9	Minimum gain of output signals
			/GAINVU	/ IDX1...I DXn	/ VALU E	R		Input level Transform given value into dB by: 20 log (256 * value / 0x7FFFFF) + 10 log (2)
			/HOLD	/VALUE		R / W	10...1000	Hold time in ms
			/ INCREASE TIME	/VALUE		R / W	10...3000	Time for GAIN INCREASE in ms

			/LINK	/VALUE		R / W	0...7	Linking of inputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-Buttons.
			/MUTE	/IDX1...IDXn		R / W	0, 1	0 = not muted, 1 = muted
			/RATIO	/VALUE		R / W	0.3...4	Ratio of program level change to ambient level change
			/RELEASE	/VALUE		R / W	10...3000	Release time in ms
			/THRESHOLD	/VALUE		R / W	-35...21	Threshold in dB
			/VU	/VALUE		R / W		Gain reduction as full scale value (32 bit) Transform given value into dB by: 20 log (0x7FFFFF/ value)

Commands for Auto mixer

					Read Write	Values	Description
/PARAM	/DSP	/AUTOMIXER_x	/AUTO	/IDX1...IDXn	R / W	0, 1	Automatic mixing of inputs
			/CLIP	/IDX1...IDXn	R	0, 1	Clipping LED of inputs and outputs
			/FREEZEGAIN	/VALUE	R / W	0, 1	Freeze gain of auto mixer
			/GAININ	/IDX1...IDXn	R / W	-80...0	Gain of inputs
			/GAINOUT	/VALUE	R / W	-80...18	Gain of output
			/POLARITYIN	/IDX1...IDXn	R / W	0, 1	Polarity of inputs (0 = normal, 1 = inverted)
			/POLARITYOUT	/VALUE	R / W	0, 1	Polarity of output (0 = normal, 1 = inverted)

			/MUTEIN	/ IDX1...IDXn	R / W	0, 1	Mute of inputs (0 = not muted, 1 = muted)
			/ MUTEOUT	/VALUE	R / W	0, 1	Mute of output (0 = not muted, 1 = muted)
			/PRIO	/VALUE	R / W	0...n	Number of input with high priority
			/ RAMPTIME	/VALUE	R / W	1...20000	Time constant (in ms) of faders
			/SIGNAL	/ IDX1...IDXn	R	0, 1	Signal LED of inputs and outputs
			/SOLO	/ IDX1...IDXn	R / W	0, 1	Solo of inputs
			/ TAUDIVDUGAN	/VALUE	R / W	1...2000	Time constant (in ms) for weighting rate of input signal level, based on total level.
			/ TAUDIVFINAL	/VALUE	R / W	1...2000	Time constant (in ms) for rate of level change.
			/ TAURMSDUGAN	/VALUE	R / W	1...2000	Time constant (in ms) of RMS measurement of dugan-gain weighted input signals.
			/ TAURMSIN	/VALUE	R / W	1...2000	Time constant (in ms) of RMS measurement of input signals.
			/VUIN	/ IDX1...IDXn	R		Input level Transform given value into dB by: 20 log (256 * value / 0x7FFFFF) + 10 log (2)
			/VUOUT	/VALUE	R	value1 value2	Output level Transform given values into dB by: 20 log (value1 / 0x7FFFFF) + value2

Commands for Compressor

					Read Write	Values	Description
/PARAM	/DSP	/COMPRESSOR_x	/ATTACK	/VALUE	R / W	5...150	Attack time in ms
			/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/GAINREDUCTION	/VALUE	R		Gain reduction as full scale value (32 bit) Transform given value into dB by: 20 log (0x7FFFFF/ value)
			/MAKE_UP_GAIN	/VALUE	R / W	-18...6	Gain of compressor output in dB, not available for RCM24 compressor
			/RATIO	/VALUE	R / W	1...100	Compressor ratio
			/RELEASE	/VALUE	R / W	10...1000	Release time in ms
			/SOFTKNEE	/VALUE	R / W	0...20	Soft knee in dB, not available for RCM24 compressor
			/THRESHOLD	/VALUE	R / W	-9...21	Threshold in dB

Commands for Delay

					Read Write	Values	Description
/PARAM	/DSP	/DELAY_x	/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated

			/ TEMPERA TURE	/VALUE	R / W	-20...60	Temperature in degree Celsius
			/VALUE	/VALUE	R / W	einheitenabh äng ig	Delay time including unit. The maximum delay time depends on the used DSP block. Available units are ms (millisecond), smp (samples), ft (Foot), inch, m (meter), cm (centimeter), us (microsecond) and s (seconds). Example: 588.235 ms

Commands for DI-1 Module

					Read Write	Values	Description
/ PARA M	/DSP	/ DIGITAL IN_x	/CHANNELINFO	/VALUE	R / W	0...127	Linking of inputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-But- tons.
			/ CHANNELMOD E	/ IDX1...IDX 4	R	0...FF	Channel mode of input signal 0 = Not indicated, 1 = 2 channel, 2 = 1 channel, 3 = primary/secondary, 4 = stereo, 5 / 6 = reserved for user applications, 7 = SCDSR, 8 = SCDSR (stereo left), 9 = SCDSR (stereo right), FF =Multichannel
			/ CHANNELSTAT USBYTES	/ IDX1...IDX 40	R		First five bytes of Channel Status Block.
			/ CLOCKACCURA CY	/ IDX1...IDX 4	R	0...3	Clock accuracy of input signal (Consumer mode only) 0 = Level 2 (+/- 1000 ppm) 1 = Level 3 (variable pitch) 2 = Level 1(+/- 50 ppm, high accuracy) 3 = reserved
			/COPYRIGHT	/ IDX1...IDX 4	R	0, 1	Copyright bit of input signal

			/ERROR	/ IDX1...IDX 4	R	00...FF	State of signal transmission (Bit[2] = Confidence Error, Bit[3] = Validity, Bit[5] = Channel Status Block CRC)
			/EMPHASIS	/ IDX1...IDX 4	R	0, 1	Emphasis bit of input signal
			/GAIN	/ IDX1...IDX 8, ALL	R / W	-80...18	Gain of input
			/LOCK	/ IDX1...IDX 4	R	0, 1	Synchronization of DI-1 input to input signal (0 = not synchronized, 1 = synchronized)
			/ LOWGROUPDE LAY	/ IDX1...IDX 4	R / W	0, 1	Low group delay option in interpolation filter of sample rate converter
			/MODE	/ IDX1...IDX 4	R	0, 1	Mode of signal transmission 0 = consumer mode 1 = Professional mode
			/MUTE	/ IDX1...IDX 8, ALL	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/ IDX1...IDX 8, ALL	R / W	0, 1	0 = normal, 1 = inverted
			/NAME	/ IDX1...IDX 8, ALL	R / W	max. 16 charact ers	Input name
			/ORIGINAL	/ IDX1...IDX 4	R	0, 1	Original bit of input signal
			/RAMPTIME	/ IDX1...IDX 8, ALL	R / W	0.001... 20	Ramping Time of faders (in seconds)
			/SAMPLERATE	/ IDX1...IDX 8	R		Sample rate of input signal
			/SRCBYPASS	/ IDX1...IDX 4	R / W	0, 1	Sample Rate Converter Bypass (0 = Bypass not activated, 1 = Bypass activated)

			/ SRCWORDLENGTH	/ IDX1...IDX4	R		Source word length of input signal (professional mode only)
			/ SOURCESELECT	/ IDX1...IDX4	R / W	1...3	Source selection of input signal 1 = AES/EBU, 2 = S/P DIF, 3 = OPTICAL

Commands for DO-1 Module

					Read Write	Values	Description
/PARAM	/DSP	/DIGITALOUT_x	/CHANNELINFO	/VALUE	R / W	0...127	Linking of outputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-Buttons.
			/GAIN	/IDX1...IDX8, ALL	R / W	-80...18	Gain of output
			/MUTE	/IDX1...IDX8, ALL	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/IDX1...IDX8, ALL	R / W	0, 1	0 = normal, 1 = inverted
			/NAME	/IDX1...IDX8, ALL	R / W	max. 16 characters	Output name
			/RAMPTIME	/IDX1...IDX8, ALL	R / W	0.001...20	Ramping Time of faders (in seconds)

Commands for Ducker

					Read Write	Values	Description
/PARAM	/DSP	/DUCKER_x	/ATTACK	/VALUE	R / W	5...1000	Attack time in ms
			/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/DUCKINGLEVEL	/VALUE	R / W	-100...-6	Ducking level in dB
			/GAINREDUCTION	/VALUE	R		Gain reduction as full scale value (32 bit) Transform given value into dB by: 20 log (0x7FFFFF/ value)
			/HOLD	/VALUE	R / W	10...2000	Hold time in ms
			/LINEMUTE	/VALUE	R / W	0, 1	Mute of LINE input (0 = not muted, 1 = muted)
			/MICMUTE	/VALUE	R / W	0, 1	Mute of MIC input (0 = not muted, 1 = muted)
			/MIXLINE	/VALUE	R / W	-30...0	Gain of LINE input
			/MIXMIC	/VALUE	R / W	-30...0	Gain of MIC input
			/RELEASE	/VALUE	R / W	5...1000	Release time in ms
			/THRESHOLD	/VALUE	R / W	-15...21	Threshold in dB

Commands for Expander

					Read Write	Values	Description
/PARAM	/DSP	/EXPANDER_x	/ATTACK	/VALUE	R / W	5...150	Attack time in ms
			/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/GAINREDUC- TION	/VALUE	R		Gain reduction as full scale value (32 bit) Transform given value into dB by: 20 log (0x7FFFFF/ value)
			/MAKE_UP_GAIN	/VALUE	R / W	-18...6	Gain of compressor output in dB, not available for RCM24 compressor
			/RATIO	/VALUE	R / W	1...10	Compressor ratio
			/RELEASE	/VALUE	R / W	10...1000	Release time in ms
			/THRESHOLD	/VALUE	R / W	-84...-25	Threshold in dB

Commands for FIR Filter

					Read Write	Values	Description
/PARAM	/DSP	/FIR_x	/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/FS	/VALUE	R	20...20000	Sampling frequency of filter
			/HIPASS.FREQ	/VALUE	R	20...20000	Frequency of high pass
			/LOPASS.FREQ	/VALUE	R	20...20000	Frequency of low pass
			/MAXORDER	/VALUE	R		Maximum order of filter
			/ORDER	/VALUE	R		Order of loaded filter
			/SLOPE	/VALUE	R	21...100	Slope of filter (in dB)
			/TYPE	/VALUE	R	0 = Low Pass, 1 = High Pass, 2 = Band Pass	Type of filter

Commands for DSP Presets

				Read Write	Values	Description
/PARAM	/DSP	/FUNCTIONS	/DELETE_PRESET	exec	1...60	Delete preset
			/LOAD_PRESET	exec	1...60	Load preset
			/PRESET_INFO_READ	exec	1...60	Read description of preset
			/PRESET_SET_DESC	W	1...60 descrip-tion	Set description of preset (max. 32 characters)
			/PRESET_SET_PROP	W	1...60 MUTE/NOMUTE	Set mute/unmute of preset during load

			/SAVE_PRESET		1...60 NOCOMP / WITHCOMP MUTE/ NOMUTE	Save preset with selectable compression and mute/ unmute during load
			/STARTUP_PRESET	exec	1...60	Set preset as startup preset
		/ACTIVEPRESET	/VALUE	R	1...60	Currently active preset

Commands for Gate

					Read Write	Values	Description
/PARAM	/DSP	/GATE_x	/ATTACK	/VALUE	R / W	5...150	Attack time in ms
			/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/GAINREDUCTION	/VALUE	R		Gain reduction as fullscale value (32 bit) Transform value into dB by: 20 log (0x7FFFFF/ value)
			/HOLD	/VALUE	R / W	5...1000	Hold time in ms
			/MAKE_UP_GAIN	/VALUE	R / W	-18...6	Gain of output
			/RELEASE	/VALUE	R / W	10...1000	Release time in ms
			/THRESHOLD	/VALUE	R / W	-84...-25	Threshold in dB

Commands for graphical Equalizer (GEQ)

					Read Write	Values	Description
/PARAM	/DSP	/GRAPHICEQ_x	/BYPASS	/IDX1...IDXn	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/LOWFREQ	/VALUE	R / W	20...1000	Frequency of LF FILTER

			/ HIGHFREQ	/VALUE	R / W	1000...20000	Frequency of HF FILTER
			/GAIN	/IDX1...IDXn	R / W	-12...12	Gain of filter. Index 1 corresponds to the LF FIL- TER, index n corresponds to the HF FILTER.
			/ QUALITY	/IDX1	R / W	0.4...40	Quality of LF FILTER
				/IDX2	R / W	3...10	Quality of filter 1 to n
				/IDX3	R / W	0.4...40	Quality of HF FILTER
			/SLOPE	/IDX1	R / W	1 = 6dB/ Oct, 2= 12dB/Oct	Slope of LF FILTER
				/IDX2	R / W	1 = 6dB/ Oct, 2= 12dB/Oct	Slope of HF FILTER
			/TYPE	/IDX1	R / W	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass	Type des LF FILTER
				/IDX2	R / W		for future use
				/IDX3	R / W	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass	Type of HF FILTER

Commands for Limiter

					Read Write	Values	Description
/PARAM	/DSP	/LIMITER_x	/ATTACK	/VALUE	R / W	0...50	Attack time in ms
			/BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/GAINREDUCTION	/VALUE	R		Gain reduction as full scale value (32 bit) Transform value into dB by: 20 log (0x7FFFFF/ value)
			/RELEASE	/VALUE	R / W	10...1000	Release time in ms
			/THRESHOLD	/VALUE	R / W	-9...21	Threshold in dB

Commands for Loudspeaker Controller

						Read Write	Values	Description
/PARAM	/DSP	/LSPKBLOCK_x	/CHn	/COMPRESSOR.ATTACK	/VALUE	R / W	5...50	Attack time in ms.
				/COMPRESSOR.BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
				/COMPRESSOR.GAINREDUCTION	/VALUE	R		Gain reduction as full scale value (32 bit) Transform value into dB by: 20 log (0x7FFFFF/ value)
				/COMPRESSOR.RATIO	/VALUE	R / W	1...8	Compressor ratio
				/COMPRES-SOR.RELEASE	/VALUE	R / W	50...999	Release time in ms
				/COMPRESSOR.THRESHOLD	/VALUE	R / W	-9...21	Threshold in dB

				/ DELAY.BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
				/EQ.BYPASS	/ IDX1...IDX6	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
				/EQ.FREQ	/ IDX1...IDX6	R / W	20...20000	Frequency of equalizer band in Hz.
				/EQ.GAIN	/ IDX1...IDX6	R / W	-18...12	Gain of equalizer band in dB
				/EQ.QUALITY	/ IDX1...IDX6	R / W	0.4...40	Quality of equalizer band
				/EQ.SLOPE	/ IDX1...IDX6	R / W	1, 2	Slope of equalizer band (1 = 6dB/Oct, 2 = 12 dB/ Oct)
				/EQ.TYPE	/ IDX1...IDX6	R / W	0...5	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass, 5 = Allpass
				/ LIMITER.ATTACK	/VALUE	R / W	0...50	Attack time in ms
				/ LIMITER.BYPASS	/VALUE	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
				/ LIMITER.GAINREDUCTION	/VALUE	R		Gain reduction as fullscale value (32 bit) Transform value into dB by: 20 log (0x7FFFFF/ value)
				/ LIMITER.RELEASE	/VALUE	R / W	10...999	Release time in ms
				/ LIMITER.THRESHOLD	/VALUE	R / W	-9...21	Threshold in dB
				/XOVER.GAIN	/VALUE	R / W	-30...6	Gain of crossover ways.

				/XOVER.MUTE	/VALUE	R / W	0, 1	0 = not muted, 1 = muted
				/ XOVER.POLARITY	/VALUE	R / W	0, 1	0 = normal, 1 = inverted
				/XOVER.LINK	/VALUE	R / W	2, 4, 6, 8 , 10, 12, 14, 16	Links the LOPASS and HIPASS of adjacent ways. Link-Numbers are used for linking two ways, for linking more than two ways the sum of the corresponding Link-Numbers is used.
				/ XOVER.LOPASS.FREQ	/VALUE	R / W	20...20000	Frequency of lopass.
				/ XOVER.LOPASS.TYPE	/VALUE	R / W	0...17	Type of lopass. 0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/ Q0.7, 5 = 12dB/ Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/ Q1.2, 8 = 12dB/ Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butterworth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB-Butterworth, 17 = 24dB-Linkwitz

				/XOVER.HIPASS.FREQ	/VALUE	R / W	20...20000	Frequency of hipass.
				/XOVER.HIPASS.TYPE	/VALUE	R / W	0...17	Type of hipass. 0 = Off, 1 = 6dB-Butter-worth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/ Q0.7, 5 = 12dB/ Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/ Q1.2, 8 = 12dB/ Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butterworth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB-Butterworth, 17 = 24dB-Linkwitz

Commands for Matrix Mixer

					Read Write	Values	Description
/PARAM	/DSP	/MATRIX_x	/CONNECTCROSSPOINT	/IDX1...IDXc	R / W	0, 1	Connection of crosspoint. Crosspoints are numbered column by column from top left to bottom right. (0 = not connected, 1 = connected)
			/GAINCROSSPOINT	/IDX1...IDXc	R / W	-80...0	Gain of crosspoint. Crosspoints are numbered column by column from top left to bot- tom right.
			/GAININ	/IDX1...IDXn	R / W	-80...0	Gain of inputs 1 to n of matrix

			/GAINOUT	/ IDX1...IDXm	R / W	-80...0	Gain of output 1 to m of matrix
			/MUTEIN	/ IDX1...IDXn	R / W	0, 1	Mute of input 1 to n of matrix
			/MUTEOUT	/ IDX1...IDXm	R / W	0, 1	Mute of output 1 to m of matrix

Commands for Mixer

					Read Write	Values	Description
/ PARAM	/DSP	/ MIXER_x	/CLIP	/IDX1...IDXn	R	0, 1	Clipping LED of inputs and outputs
			/GAININ	/IDX1...IDXn	R / W	-80...0	Gain of inputs
			/ GAINOUT	/IDX1	R / W	-80...18	Gain of output L (only for stereo mixer)
				/IDX2	R / W	-80...18	Gain of output R
			/LINK	/VALUE	R / W	0x00...0xFF	Linking of inputs (Linked-Button). Binary representation of number is equivalent to pressed Linked-Buttons.
			/MUTEIN	/IDX1...IDXn	R / W	0, 1	Mute of inputs (0 = not muted, 1 = muted)
			/ MUTEOUT	/IDX1	R / W	0, 1	Mute of output L (only for stereo mixer)
				/IDX2	R / W		Mute of output R
			/PAN	/IDX1...IDXn	R / W	-50...+50	Pan of input (only for stereo mixer, -50 = left, +50 = right)
			/ POLARITY	/IDX1...IDXn	R / W	0, 1	0 = normal, 1 = inverted
			/SIGNAL	/IDX1...IDXn	R	0, 1	Signal LED of inputs and outputs
			/SOLO	/IDX1...IDXn	R / W	0, 1	Solo of inputs

Commands for Noise Generator

					Read Write	Values	Description
/PARAM	/DSP	/NOISE GENERATOR_x	/ENABLE	/VALUE	R / W	0, 1	0 = off, 1 = on
			/GAIN	/VALUE	R / W	-60...0	Gain of noise in dB
			/TYPE	/VALUE	R / W	0, 1	0 = white noise, 1 = pink noise

Commands for parametric Equalizer (PEQ)

					Read Write	Values	Description
/PARAM	/DSP	/PEQ_x	/BYPASS	/IDX1...IDXn	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/FREQ	/IDX1...IDXn	R / W	20...20000	Frequency in Hz
			/GAIN	/IDX1...IDXn	R / W	-18...12	Gain of equalizer band
			/QUALITY	/IDX1...IDXn	R / W	0.4...40	Quality of equalizer band
			/SLOPE	/IDX1...IDXn	R / W	1, 2	Slope of equalizer band (1 = 6dB/Oct, 2 = 12dB/Oct)
			/TYPE	/IDX1...IDXn	R / W	0,1,2,3,4,5	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass, 5 = Allpass

Commands for Priority Matrix

					Read Write	Values	Description
/PARAM	/DSP	/PRIORITYMATRIX_x	/CONNECTPAGINGLINE	/IDX1...IDXn	R / W	see Description	Connection of crosspoint. Crosspoints are numbered column by column from top left to bottom right. (0 = not connected, 1 = connected)
			/GAINCROSSPOINT	/IDX1...IDXn	R / W	-80...0	Gain of crosspoint. Crosspoints are numbered column by column from top left to bottom right.
			/GAINOUT	/IDX1...IDXm	R / W	-80...0	Gain of outputs.
			/MUTEIN	/IDX1...IDXn	R / W	0, 1	Mute of inputs (0 = not muted, 1 = muted)
			/MUTEOUT	/IDX1...IDXm	R / W	0, 1	Mute of outputs (0 = not muted, 1 = muted)
			/PRIORITYPAGINGLINE	/IDX1...IDXn	R / W	0...255	Priority of inputs

Commands for Router

					Read Write	Values	Description
/PARAM	/DSP	/ROUTER_x	/ROUTING POINTS	/IDX1...IDXm	R / W	0...n	Connected crosspoint of output 1 to m of router

Commands for Tone Control

					Read Write	Values	Description
/PARAM	/DSP	/TONE CONTROL_x	/GAIN	/ IDX1...IDX3	R / W	-18...12	Gain of band

Commands for Tone generator

					Read Write	Values	Description
/ PARAM	/DSP	/TONE GENERATOR_x	/ENABLE	/VALUE	R / W	0, 1	0 = off, 1 = on
			/FREQ	/VALUE	R / W	20...20000	Constant frequency (for SINE mode)
			/GAIN	/VALUE	R / W	-60...0	Gain of generated tone
			/SWEEPREP	/VALUE	R / W	0, 1	Selection of a single or periodic sweep (0 = one time, 1 = periodic)
			/SWEEPSTART	/VALUE	R / W	20...20000	Lower (or upper) frequency of sweep
			/SWEEPSTOP	/VALUE	R / W	20...20000	Upper (or lower) frequency of sweep
			/SWEEPTIME	/VALUE	R / W	5...120	Time of sweep in seconds
			/TYPE	/VALUE	R / W	0, 1	Selection of SINE or SWEEP mode (0 = SINE, 1 = SWEEP)

Commands for Crossover

					Read Write	Values	Description
/PARAM	/DSP	/ XOVER _x	/GAIN	/ IDX1...IDXn	R / W	-30...6	Gain of crossover ways. Ways are numbered top-down. (e.g. IDX1 = HIGH, IDX2 = MID, IDX3 = LOW)
			/ HIPASS.FREQ	/ IDX1...IDXn	R / W	20...20000	Frequency of hipass. Ways are numbered top-down. (e.g. IDX1 = HIGH, IDX2 = MID, IDX3 = LOW)

			/ HIPASS.TYPE	/ IDX1...IDXn	R / W	0...17	Type of hipass. 0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/Q0.7, 5 = 12dB/Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/Q1.2, 8 = 12dB/Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butter- worth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB- Butterworth, 17 = 24dB-Linkwitz
			/LINK	/VALUE	R / W	2, 4, 6, 8 , 10, 12, 14, 16	Links the LOPASS and HIPASS of adjacent ways. Link- Numbers are used for linking two ways, for linking more than two ways the sum of the corresponding Link-Numbers is used.
			/ LOPASS.FREQ	/ IDX1...IDXn	R / W	20...20000	Frequency of lopass. Ways are numbered top-down. (e.g. IDX1 = HIGH, IDX2 = MID, IDX3 = LOW)
			/ LOPASS.TYPE	/ IDX1...IDXn	R / W	0...17	Type of lopass. 0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/Q0.7, 5 = 12dB/Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/Q1.2, 8 = 12dB/Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butter- worth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB- Butterworth, 17 = 24dB-Linkwitz
			/MUTE	/ IDX1...IDXn	R / W	0, 1	0 = not muted, 1 = muted
			/POLARITY	/ IDX1...IDXn	R / W	0, 1	0 = normal, 1 = inverted

Commands for RCM-24

					Read Write	Values	Description
/PARAM	/RCM24	/COMMON	/CANBAUDRATE	/VALUE	R	10, 20, 63, 125, 250, 500	Baud rate of CAN-BUS
			/AMPNAME	/IDX1...IDX250	R / W	max. 30 characters	Amplifier name
			/CONFIG	/VALUE	R		List of configured amplifiers
			/POWER	/IDX1...IDX250	R / W	0, 1	Switch amp ON / OFF or read out ON / OFF state (0 = off, 1 = on)
			/POWERDELAY	/IDX1...IDX250	R / W	1...127	Power-On-Delay in steps of 20ms. 0 sets the default value, dependent on amp address.
			/THERMO	/IDX1...IDX250	R		Current amplifier temperature in degree Celsius
			/THERMORANGE	/IDX1...IDX250	R / W	20...150, 0...40	Temperature limit and hysteresis for OVER TEMPERATURE flag in degree Celsius
		/FUNCTIONS	/LOAD_PRESET		exec	1...8 "amps"	Loads a amplifier preset. 1 = U01, 2 = U02,..., 8 = U08; "amps" is hexadecimal representation of amps.
			/SAVEPRESET		exec	1...8 „amps)	Saves a amplifier preset. 1 = U01, 2 = U02,..., 8 = U08; "amps" is hexadecimal representation of amps.
		/INFO			R		Information about connected RCM-24 amplifiers
		/INPA	/NAME	/IDX1...IDX250	R / W	max. 30 characters	Name of input A
			/DELAYBYPASS	/IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/DELAYVALUE	/IDX1...IDX250			
			/EQ1BYPASS	/IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated

			/EQ1TYPE	/ IDX1...IDX250	R / W	0, 1 ,2, 3, 4	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass
			/ EQ1SLOPE	/ IDX1...IDX250	R / W	1, 2	Slope of equalizer 1 (1 = 6dB/Oct, 2 = 12dB/Oct)
			/ EQ1FREQ	/ IDX1...IDX250	R / W	20...20000	Frequency of equalizer 1 in Hz
			/EQ1GAIN	/ IDX1...IDX250	R / W	-18...12	Gain of equalizer 1
			/ EQ1QUALITY	/ IDX1...IDX250	R / W	0.4...40	Quality of equalizer 1
			/EQ2...				(same as above, but for equalizer 2 to 5)
			...				
			/EQ5...				
		/INPB	...				(same as above, but for input B)
		/OUTPA	/NAME	/ IDX1...IDX250	R / W	max. 30 characters	Name of output A
			/LEVEL	/ IDX1...IDX250	R / W	-128...6	Gain of output A
			/ TRIMLEVEL	/ IDX1...IDX250	R / W	-30...6	Gain Trim (in dsp block Crossover)
			/ DELAYBYPASS	/ IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/ DELAYVALUE	/ IDX1...IDX250			
			/MUTE	/ IDX1...IDX250	R / W	0, 1	0 = not muted, 1 = muted
			/ POLARITY	/ IDX1...IDX250	R / W	0, 1	0 = normal, 1 = inverted
			/ROUTE	/ IDX1...IDX250	R / W	0, 1, 2	Routing of output A (0 = A, 1 = B, 2 = A+B)
			/ COMPBYPASS	/ IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated

			/ COMPTYPE	/ IDX1...IDX250	R / W	0, 1, 2, 3, 4	Compressor ratio (0 = 1/1, 1= 1/1.4, 2 = 1/2, 3 = 1/4, 4 = 1/8)
			/ COMPTHRESH	/ IDX1...IDX250	R / W	-30...0	Threshold of compressor in dB (-30 corresponds to -9 dB, 0 corresponds to +21 dB)
			/ COMPATTACK	/ IDX1...IDX250	R / W	0...99	Attack time of compressor in ms
			/ COMPRELEASE	/ IDX1...IDX250	R / W	10...999	Release time of compressor in ms
			/ LIMITBYPASS	/ IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/ LIMITTHRES	/ IDX1...IDX250	R / W	-30...0	Threshold of limiter in dB (-30 corresponds to -9 dB, 0 corresponds to +21 dB)
			/ LIMITRELEASE	/ IDX1...IDX250	R / W	10...999	Release time of limiter in ms
			/ XOVERHITYPE	/ IDX1...IDX250	R / W	0...17	0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/Q0.7, 5 = 12dB/Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/Q1.2, 8 = 12dB/Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butterworth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB-Butterworth, 17 = 24dB-Linkwitz
			/ XOVERHIFREQ	/ IDX1...IDX250	R / W	20...20000	Frequency of crossover-hipass of output A
			/ XOVERLOTYPE	/ IDX1...IDX250	R / W	0...17	0 = Off, 1 = 6dB-Butterworth, 2 = 12dB/Q0.5, 3 = 12dB/Q0.6, 4 = 12dB/Q0.7, 5 = 12dB/Q0.8, 6 = 12dB/Q1.0, 7 = 12dB/Q1.2, 8 = 12dB/Q1.5, 9 = 12dB/Q2.0, 10 = 12dB-Bessel, 11 = 12dB-Butterworth, 12 = 12dB-Linkwitz, 13 = 18dB-Bessel, 14 = 18dB-Butterworth, 15 = 24dB-Bessel, 16 = 24dB-Butterworth, 17 = 24dB-Linkwitz

			/ XOVERLOFREQ	/ IDX1...IDX250	R / W	20...20000	Frequency of crossover-lopass of output A
			/ EQ1BYPASS	/ IDX1...IDX250	R / W	0, 1	0 = Bypass not activated, 1 = Bypass activated
			/EQ1TYPE	/ IDX1...IDX250	R / W	0, 1, 2, 3, 4, 5	0 = PEQ, 1 = Loshelv, 2 = Hishelv, 3 = Hipass, 4 = Lopass, 5 = Allpass
			/ EQ1SLOPE	/ IDX1...IDX250	R / W	1, 2	Slope of equalizer 1 (1 = 6dB/Oct, 2 = 12dB/Oct)
			/ EQ1FREQ	/ IDX1...IDX250	R / W	20...20000	Frequency of equalizer 1 in Hz
			/EQ1GAIN	/ IDX1...IDX250	R / W	-18...12	Gain of equalizer 1
			/ EQ1QUALITY	/ IDX1...IDX250	R / W	0.4...40	Quality of equalizer 1
			/EQ2...	/ IDX1...IDX250			(same as above, but for equalizer 2 to 5)
			...				
			/EQ5...				
		/OUTPB	...				(same as above, but for output B)

Commands for RCM-810

					Read Write	Values	Description
/PARAM	/AMP	/ADR1...250	/COMMON	/REVISION	R		Firmware version of RCM-810
				/POWER	R / W	0, 1	Switch amp ON / OFF or read out ON / OFF state (0 = off, 1 = on)
				/STATEFLAGS	R	0x00...0xFF, 0x00...0xFF, 0x00...0xFF, 0x00...0xFF	32 Bit state flags of the amplifier in hexadecimal format. 1: delayed power switch 2: standby state 3: polling timeout from CAN master 4: error in non volatile or tag memory 5: collection result of local stateflags 6: global state of monitored collect stateflags 7: overtemp (in any channel) 8 to 16 : unused 17: routing switch 'bridged' instead of 'normal' 18: mode switch 'parallel' instead of 'dual' 19 to 21: unused 22: thermal shutdown flag from atmel 23 to 26: unused 27: mains voltage warning 28 to 32: unused
				/COLLECT	R	0x00...0xFF, 0x00...0xFF, 0x00...0xFF, 0x00...0xFF	Bit mask for state flag selection in hexadecimal format.
				/GLBERR	R	0x00...0xFF, 0x00...0xFF	Global error state of external can devices, 1 bit per CAN device (1=error, 0=OK)
				/GPISTATE	R	0, 1 per GPI	State of the two GPIs. The value 0 means „not activated“, so the input is open (high impedance). The value 1 means „activated“, so the input is connected (low impedance) to signal earth.

				/GPOSTATE	R / W	0, 1 per GPO	State of the two GPOs. The value 0 means „not activated“ (high impedance). The value 1 means the output is „activated“, so the out- put is connected (low impedance) to signal earth.
				/MASTERERROR	R	0, 1	Current COLLECTED ERROR STATE of the RCM-810 (0 = not active, 1 = active)
			/IN1...8	/NAME	R	max. 30 characters	Description (name) of input channel
			/ OUT1... 8	/NAME	R	max. 30 characters	Description (name) of output channel
				/STATEFLAGS	R	0x00...0xFF, 0x00...0xFF, 0x00...0xFF	24 Bit state flags of the output channel in hexadecimal format. : SMPS reports overload : amp reports excessive Hf level : low impedance load or short-circuit detect 4 : thermal protection active 5 : in general: protect-mode active 6 : protection mute active 7 : amp output relay off 8 : unused 9 : protection gain reduction active 10 and 11: unused 12 : load value result valid 13 : speakerload shorted 14 : speakerload opened 15 : pilot not detected : unused : thermal headroom below minimum
				/COLLECT		0x00...0xFF, 0x00...0xFF, 0x00...0xFF	Bit mask for state flag selection in hexadecimal format.
				/MUTE	R / W	0, 1	0 = not muted, 1 = muted

Link-Numbers for linking adjacent ways of a crossover

Number of crossover ways	**5**	**4**	**3**	**2**	**1**
Link-Number		8		2	
	16		4		

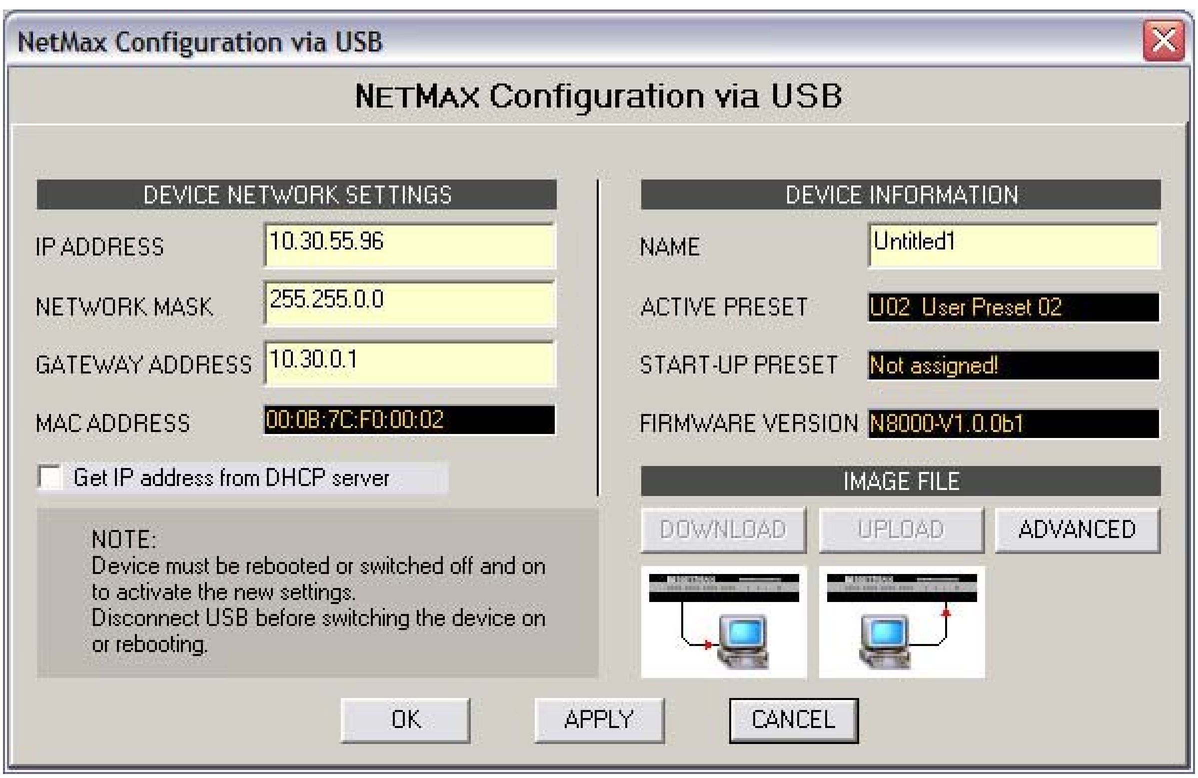
Hexadecimal representation of remote amplifiers connected to the device

CAN Address	1	2	3	4	5	6	7	8	9	...	246	247	248	249	250
Decimal value	8	4	2	1	8	4	2	1	8	...	4	2	1	8	4
Example			X	X	X		X								
	Ox3A														

Configuration via USB

Apart from the Ethernet network interface the N8000 also supports a USB interface. The USB driver is located in the \IRIS-Net\Driver\USB Netmax Driver folder and must be installed before the USB interface can be used. The N8000 can then be configured using the USB interface by selecting the Configuration via USB menu item in the Netmax menu of IRIS-Net.

HINT: To avoid configuration problems when using the USB interface only one NetMax N8000 should be connected at a time. The use of a USB hub to connect several N8000's simultaneously will result in device conflicts and is not recommended.

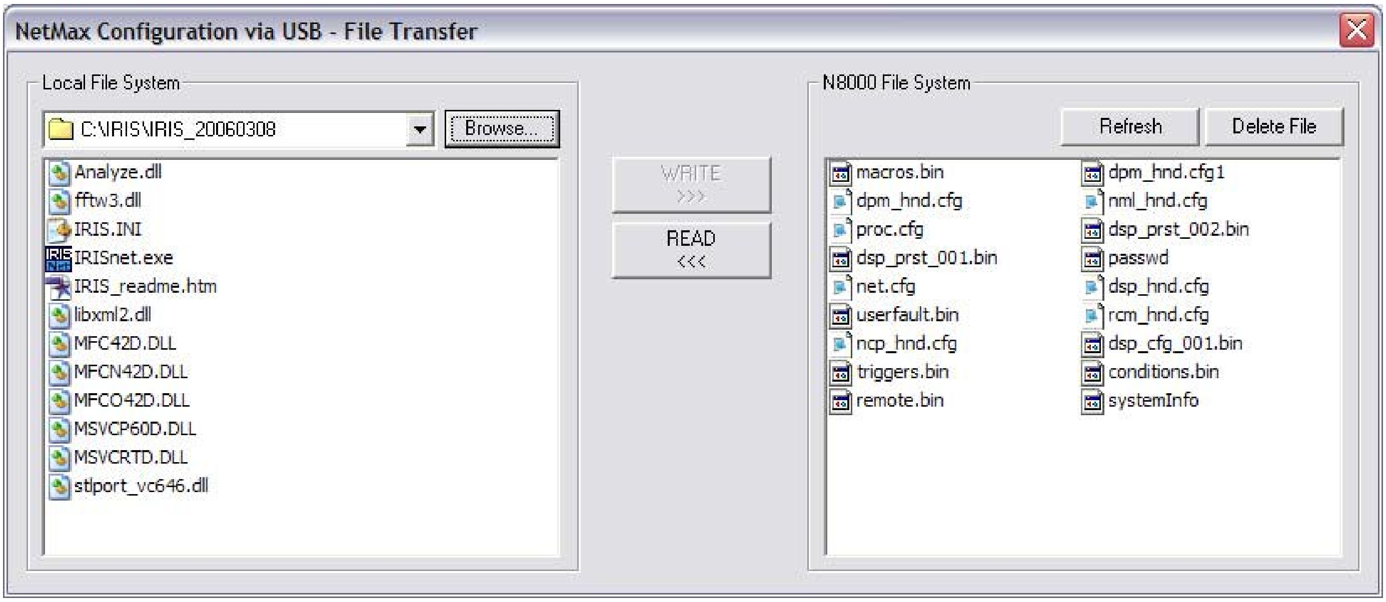


Element	Description
IP ADDRESS	Displays the IP address of the N8000 Ethernet interface (factory default: 192.168.1.100). To ensure trouble free communication with IRIS-Net each N8000 on the network must have a unique IP address.
NETWORK MASK	Setting and indication of the network mask for the Ethernet interface. (factory default: 255.255.255.0)

GATEWAY ADDRESS	Setting and indication of the Standard Gateway for the Ethernet interface. (factory default: 192.168.1.1)
MAC ADDRESS	Displays the MAC address of the connected N8000. The MAC address of the N8000 can also be found on a label attached to the rear panel of the device.
Get IP address from DHCP server	If the Ethernet network contains a DHCP server, it can be used to automatically and conveniently configure the N8000’s Ether- net interface.
NAME	Internal IRIS-Net name of the N8000.
ACTIVE PRESET	Name of the currently loaded preset. Changing presets is possible via the Preset Manager.
START-UP PRESET	Name of the currently selected start-up preset. Changing presets is possible via the Preset Manager.
FIRMWARE VERSION	Displays the firmware version of the N8000.
DOWNLOAD	Allows an Image File of the entire operating system configuration to be downloaded from an N8000 to a PC. This function is typically used to create a backup of the N8000's configuration but uploading the image file to other N8000's also provides a convenient way to transfer settings to a number of identically configured devices.
UPLOAD	Allows an Image File to be uploaded from a PC to an N8000. Typically used to restore a previously saved backup file.
ADVANCED	Opens the File Transfer dialog box used to transfer individual operating system files between the N8000 and PC.

FILE TRANSFER
The File Transfer dialog window allows experienced users to transfer individual operating system files between the N8000 and PC.

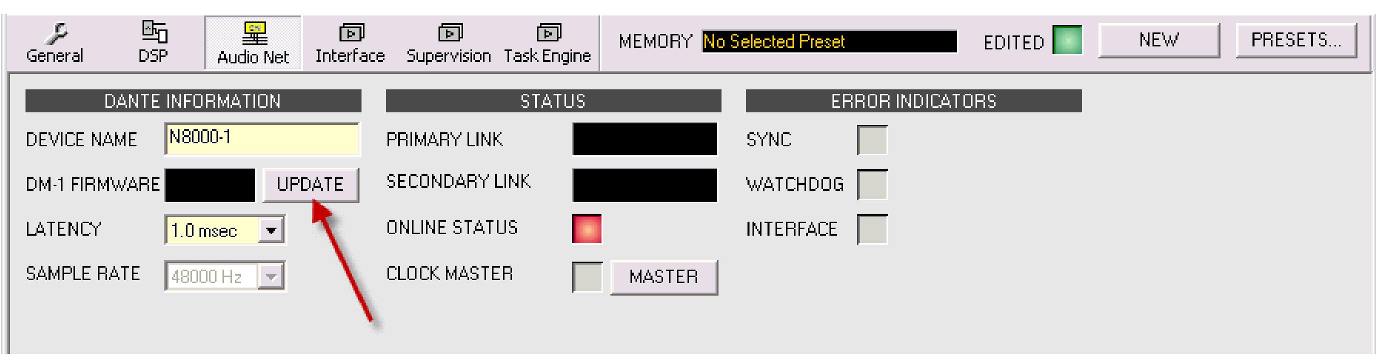
Caution!
File Transfer is an advanced operation which should only be carried out by experienced users who are fully conversant with the N8000 file system. Deleting or replacing critical N8000 operating system files can lead to malfunction and/or damage to the device. If in doubt DO NOT ATTEMPT TO USE THIS FUNCTION!
Consequences



DM-1 Firmware Update

For updating the DM-1 Dante Network Module firmware the web interface of the module is used. Execute following steps to update the firmware:

1. Connect the DM-1 to be updated to your PC via Ethernet.
2. Power on the N8000 or P 64.
3. Start the IRIS-Net application.
4. Drag & drop a N8000 or P 64 to the project and add a DM-1 module in the General dialog of the N8000 or P 64.
5. Open the Audio Net dialog and click on the UPDATE button. The web interface of the DM-1 is opened in your browser.
   HINT: If the web interface of the DM-1 does not open, go online with option “Send All to Selected Devices” to set the DEVICE NAME of the DM-1. After this, press the UPDATE button again.



6. Click on the Upgrade Firmware button in the DM-1 Web Interface.
   An open file dialog appears.
7. Open the folder /Firmware/DM-1 in the installation directory of IRIS-Net.
8. Select the file webupd-dlm-x-y-z.dnt (x, y and z are optional version numbers) and start the update process.
9. Power cycle the N8000 or P 64 after the update has finished.
10. Reload the DM-1 Web Interface in the browser after the power cycle.
11. Click on the Upgrade Firmware button in the DM-1 Web Interface.
12. Open the folder /Firmware/DM-1 in the installation directory of IRIS-Net.
13. Select the file DLM-failsave-update.dnt and start the update process.

14. Click on the Upgrade Firmware button in the DM-1 Web Interface.
15. Open the folder /Firmware/DM-1 in the installation directory of IRIS-Net.
16. Select the file NetMax_N8000-capability-update.dnt and start the update process.
17. Power cycle the N8000 or P 64 after the update has finished.
18. The DM-1 Dante Network Module can now be used.

HINT: The DM-1 Firmware Update fails if a DM-1 module with a firmware version older than V3.4.2 shall be updated to the newest Version V3.4.3-RC2. A complete firmware update to V3.4.2 (including failsafe & capability image) has to be done before the module is updated to the newest firmware version V3.4.3.

N8000 Browser Interface

The browser interface of the N8000's integrated web server is part of IRIS-Net's existing possibilities for configuring and operating the N8000. The use of a current Internet Browser is recommended to access the N8000 browser interface. The following Internet Browsers are qualified to fulfill this task:

- Microsoft Internet Explorer
- Mozilla Firefox (recommended)

HINT: Java Script and CSS have to be activated in the Internet Browser.

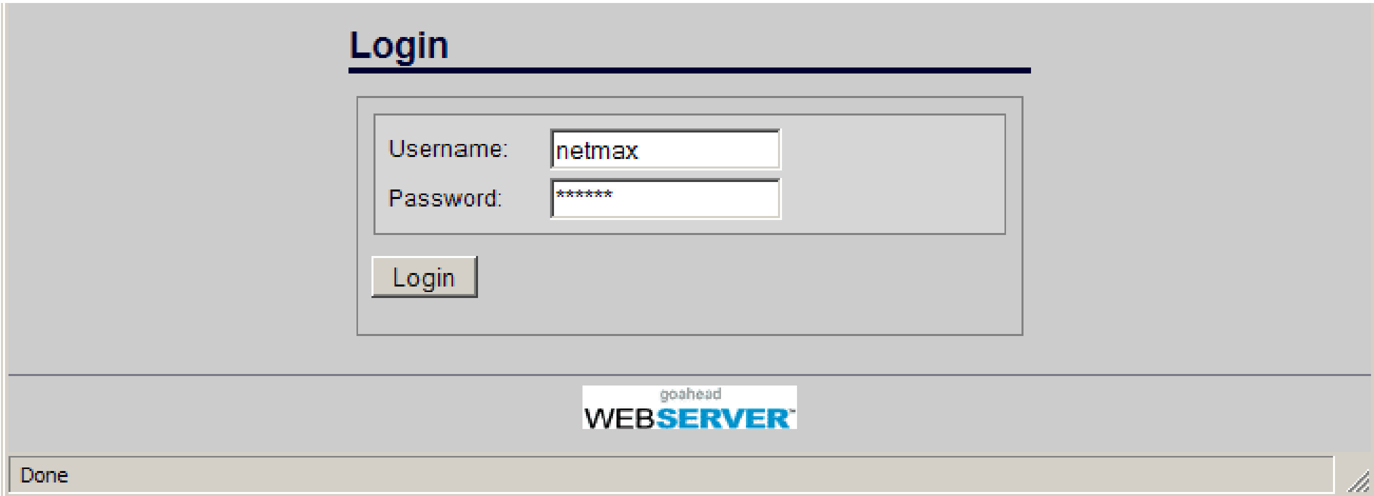
Enter the N8000's IP-address in the Internet Browser of your choice to access the browser interface. The Login page appears. The following table lists the default IP address (next to other information for configuring the network) of the NetMax N8000 System Controller.

Parameter	**Value**
IP address	192.168.1.100
Subnet mask	255.255.255.0
Gateway	192.168.1.1
DHCP	deactivated

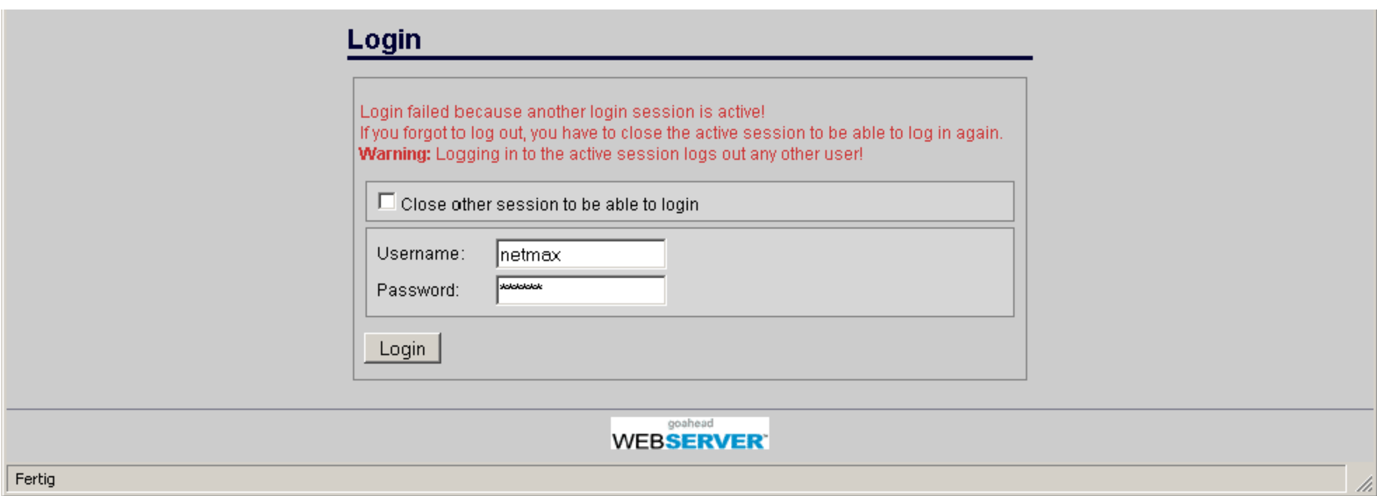
LOGIN

The browser interface login page allows users to enter username and associated password, where the factory-set password corresponds to the actual firmware version installed in the N8000 (as listed in the table below). Only one user at a time will be able to access the N8000 browser interface..

Parameter		**Value**
Username		netmax
Passwort	firmware < V0.17.0a1	nmuser
	firmware >= V0.17.0a1	netmax



The main window System Settings appears upon entering the correct login information and pressing the Login button. As shown in the screenshot below, a warning appears when you try to log in while another user already employs the browser interface. Completing the login at this point will exclude (log out) any other user from the active session.



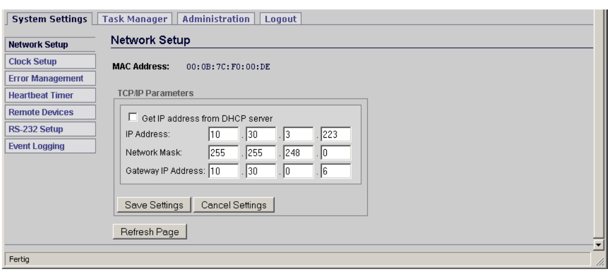
SYSTEM SETTINGS

The N8000 Browser Interface is divided into three main windows, System Settings, Task Manager and Administration. Menu entries of the System Settings window are listed in the following table:

Menu entry	Description
Network Setup	Network configuration settings of the Ethernet port
Clock Setup	Date/Time settings and automatic DST-switching
Error Management	Error status of the N8000 and error types configuration
Heartbeat Timer	Activating/Deactivating the Heartbeat Timer
RS-232 Setup	Configuration of the two RS-232 ports
Event Logging	Selecting the event types to be included in the Event Log

Network Setup

The Network Setup window allows the configuration of the N8000's Ethernet port.



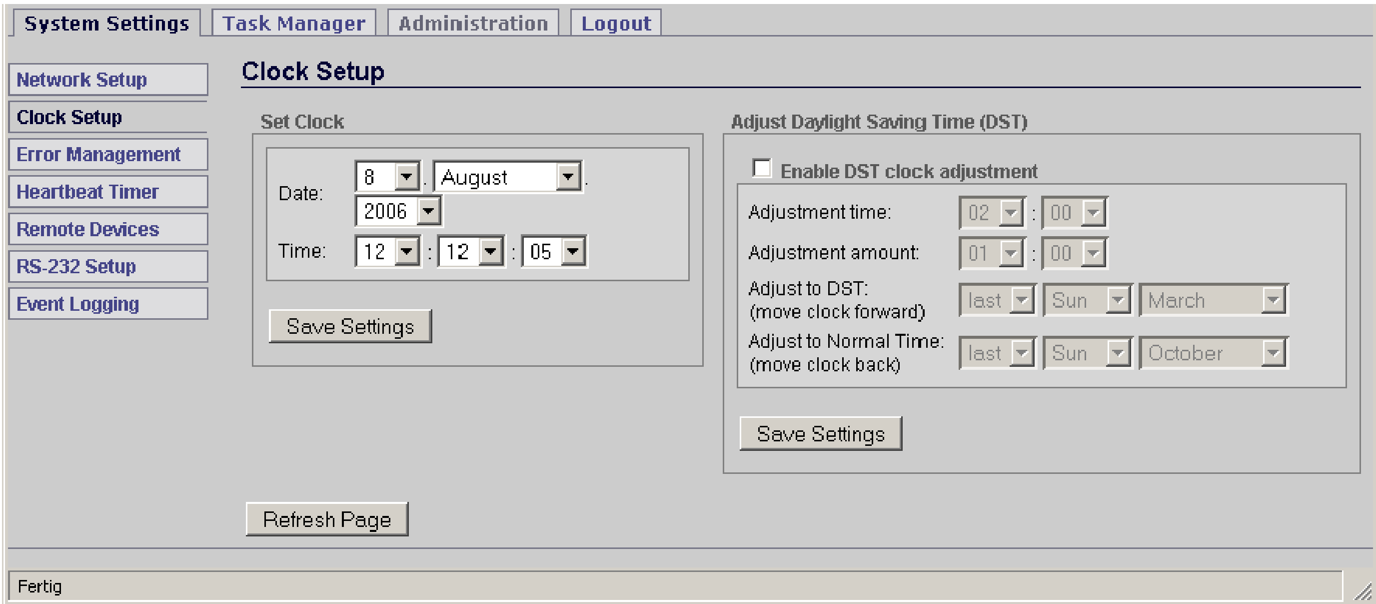
The following table shows the default network configuration of the N8000:

Parameter	Value
IP address	192.168.1.100
Subnet mask	255.255.255.0
Gateway	192.168.1.1
DHCP	deactivated

Clock Setup

The Clock Setup page allows setting the date and time of the N8000's system clock. When enabling automatic DST-switching you also have to specify two dates on which the time will be shifted. Whether adjusting the clock takes place when enabling automatic DST-switching depends on the current date and the set switching dates.

HINT: CEST - Central European Summer (Daylight Saving) Time - starts on the last Sunday in March when at 2:00 AM CET clocks are forwarded one hour to 3:00 AM CEST. CEST ends on the last Sunday in October when at 3:00 AM CEST clocks are set back one hour to 2:00 AM CET.



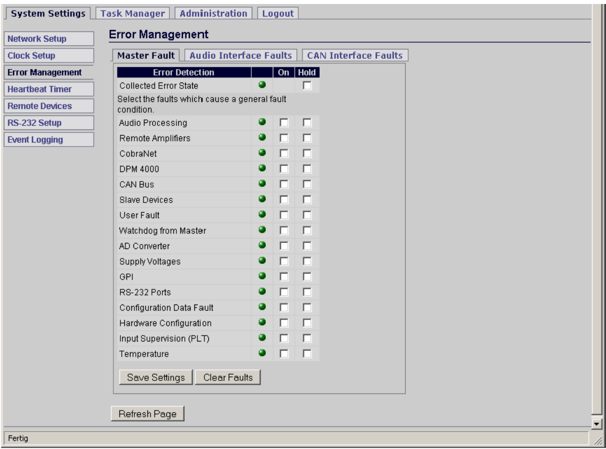
Element	Description
Adjustment time	Specifies the point in time at which the clock is forwarded when DST starts. When DST ends, the clock is set back to this time value.
Adjustment amount	Specifies the period of time by which the clock is shifted at the begin/end of DST.
Adjust to DST (move clock forward)	Specifies the date on which, at the specified Adjustment Time and by the specified Adjustment Amount, automatic STANDARD-to-DST shifting takes place.
Adjust to Normal Time (move clock back)	Specifies the date on which automatic DST-to-STANDARD shifting takes place. Time shifts back by the specified Adjustment Amount to the specified Adjustment Time.

Error Management

The Error Management page shows an actual overview of all types of NetMax N8000 errors. The current status for each type of error is displayed. Setting On and Hold options is possible.

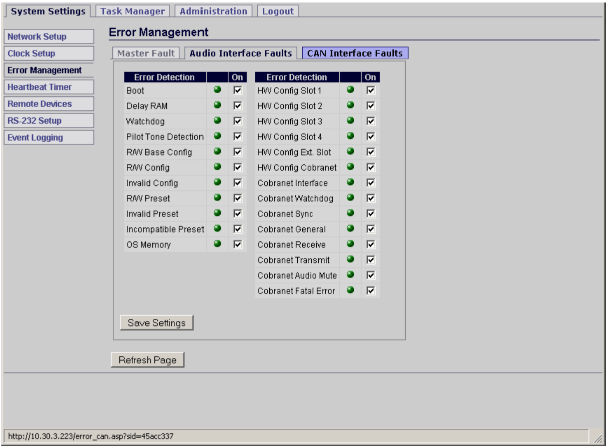
Option	Description
On	The FAULT-LED on the N8000's front panel (see also Collected Error State) signals any occurrence of errors for which the On option has been selected. In addition, all errors are logged with date and time stamps in the N8000 Log File.
Hold	The signaling of errors for which the Hold option has been selected is not reset, even if they occur only temporarily. This option is especially useful for detecting sporadic network errors.

Master Fault



Error	**Description**
Collected Error State	The FAULT-LED on the front panel of the N8000 lights at the occurrence of this error.
Audio Processing	General DSP fault, please refer to the table DSP Faults for further detail
Remote Amplifiers	General Remote Amplifier fault, please refer to the table RCM Faults for further detail
CobraNet	Error within the CobraNet-System (CM-1)
DPM 4000	The DPM 4000 connected to the RS-232 port cannot be accessed
CAN Bus	General CAN Bus error, please refer to the table RCM Faults for further detail
Slave Devices	A Remote Device signals an error or cannot be accessed
User Fault	The Error Code of at least one User Fault has been set to ≠ 0
Watchdog from Master	The Heartbeat Timer‘s time limit has been exceeded
AD Converter	Malfunction in the A/D converter of control inputs (GPI)
Supply Voltages	Error in the N8000’s internal power supply
GPI	The input voltage of a control input (GPI) is to low/high
RS-232 Ports	A unit connected to the RS-232 shows a malfunction
Configuration Data Fault	Memory/Configuration data error, please refer to tables DSP Faults and RCM Faults for further detail
Hardware Configuration	Hardware Configuration fault in the N8000, please refer to table DSP Faults for further detail
Input Supervision (PLT)	Fault in Pilot Tone Detection
Temperature	N8000 thermal overload

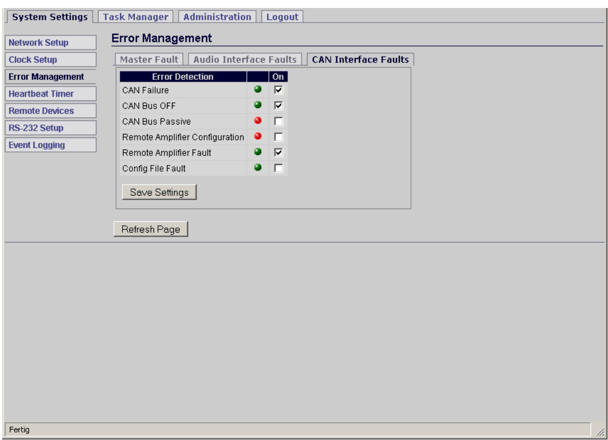
Audio Interface Faults



Error	Description
Boot	Booting the DSP system failed, Audio Processing is not possible. Please refer to the correspondent Log File entries
Delay RAM	Delay RAM fault. Audio Processing using the concerned RAM areas is not possible. Please refer to the correspondent Log File entries
Watchdog	One processor of the DSP system signals the occurrence of an error. Please refer to the correspondent Log File entries
Pilot Tone Detection	At least one of the pilot tone detections in the N8000's DSP configuration is not able to detect the pilot signal
R/W Base Config	Error while reading or writing the DSP Base Configuration
R/W Config	Error while reading or writing the DSP Configuration
Invalid Config	Unable to process invalid DSP Configuration
R/W Preset	Error while reading or writing a DSP Preset
Invalid Preset	Unable to process an invalid DSP Preset
Incompatible Preset	Selected DSP Preset and current DSP Configuration are not compatible
OS Memory	An error occurred when the operating system of the N8000 tried to access the memory
HW Config Slot 1	The DSP configuration of Audio Slot 1 does not represent the actual hardware configuration
HW Config Slot 2	The DSP configuration of Audio Slot 2 does not represent the actual hardware configuration
HW Config Slot 3	The DSP configuration of Audio Slot 3 does not represent the actual hardware configuration
HW Config Slot 4	The DSP configuration of Audio Slot 4 does not represent the actual hardware configuration
HW Config Ext. Slot	The DSP configuration requires an extension board connected to the extensions port, but there is no board connected
HW Config Cobranet	The DSP configuration requires a CobraNet board installed in the network module slot, but there is no board installed
Cobranet Interface	An error occurred during communication with the CM-1 interface

Cobranet Watchdog	A hardware or software error causes a reset of the CM-1
Cobranet Sync	Synchronizing the DSP system with CobraNet is not possible
Cobranet General	System fault in the CM-1 module
Cobranet Receive	An error occurred while receiving data from CobraNet.
Cobranet Transmit	An error occurred while sending data via CobraNet.
Cobranet Audio Mute	Audio output has been muted, because assuring correct transmission was in question.
Cobranet Fatal Error	A fatal error occurred in the CM-1

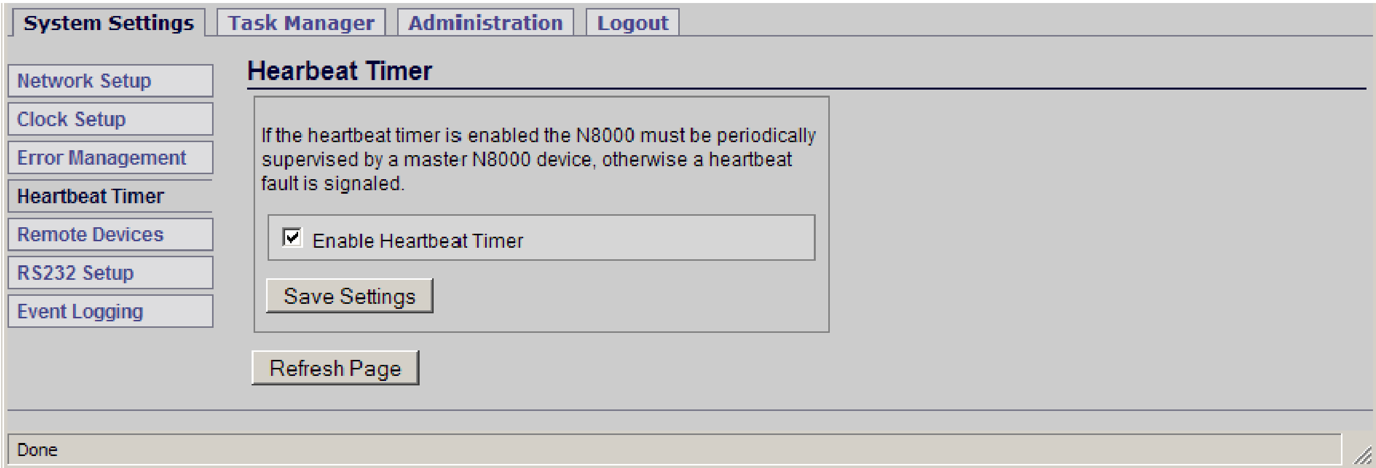
CAN Interface Faults



Error	Description
CAN Failure	CAN self-testing was not successful. The CAN-Bus does not function.
CAN Bus OFF	The CAN-Bus is in the "Bus OFF" state.
CAN Bus Passive	The CAN-Bus is in "Passive" mode.
Remote Amplifier Configuration	The RCM configuration does not represent the RCMs that are actually connected.
Remote Amplifier Fault	Collected Error for at least one RCM has been set.
Config File Fault	An error occurred while reading the Configuration File.

Heartbeat Timer

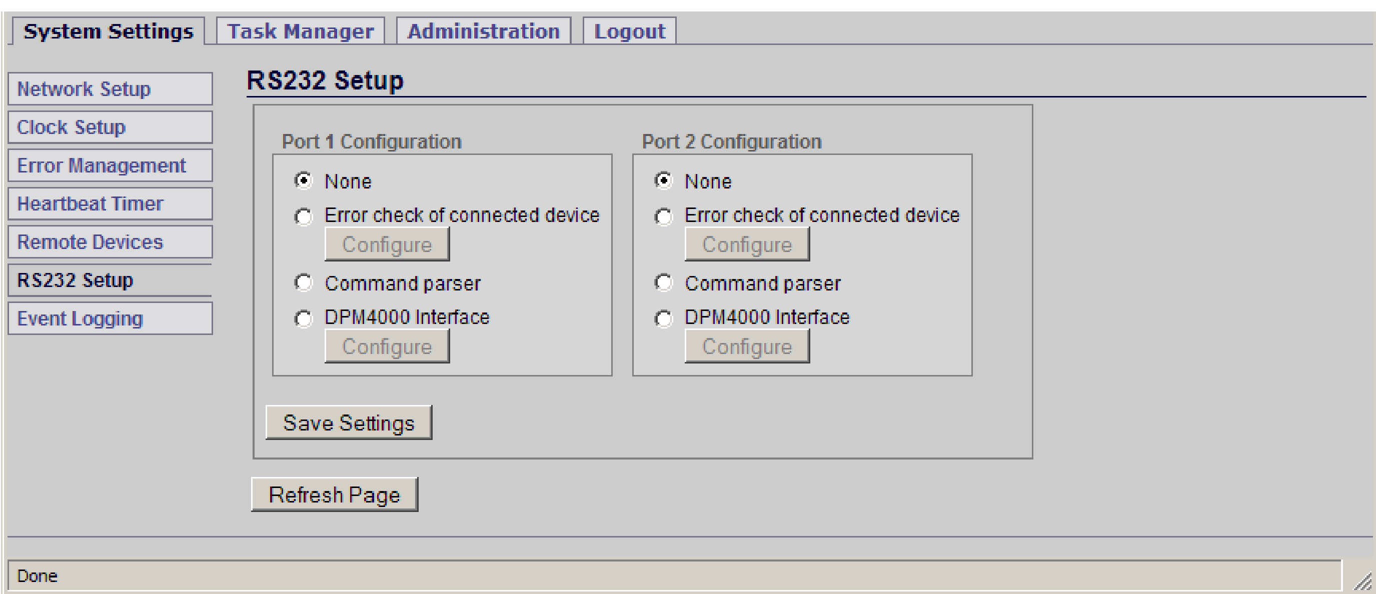
If the Heartbeat Timer of a N8000 (from here on referred to as "N8000 slave unit") has been activated, another N8000 (from here on referred to as "N8000 master unit") has to monitor/poll the slave unit. Therefore, it is necessary to enter the N8000 slave unit in the N8000 master unit’s list of Remote Devices, so that the N8000 master unit periodically sends a request via Ethernet which the N8000 slave unit has to answer. The N8000 slave unit signals a Heartbeat Fault whenever the periodic request has not been received while the N8000 server, on the other hand, signals a Remote N8000 Fault.



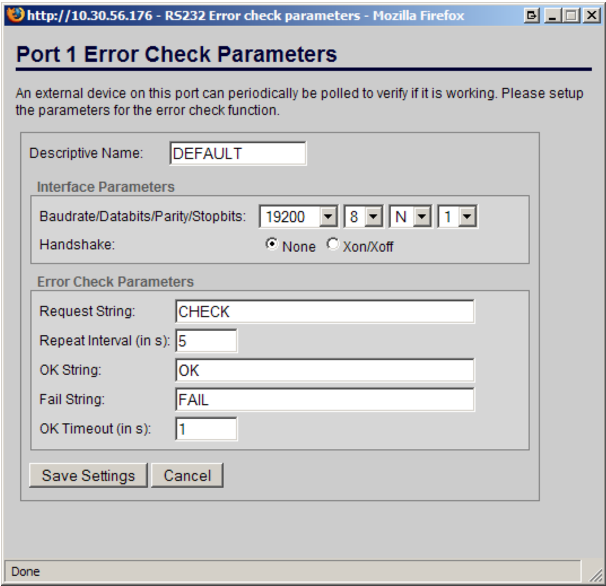
RS-232 Setup

The two RS-232 ports of a N8000 can be used to monitor other devices, for sending and receiving commands and for the connection of a ProMatrix/ProAnnounce DPM 4000.

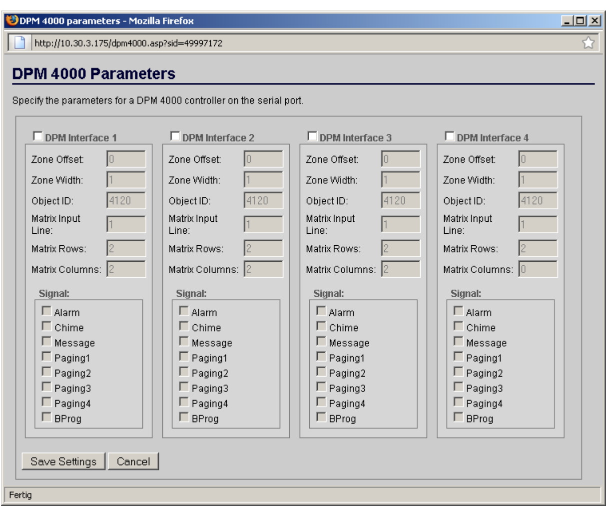
HINT: Simultaneous utilization of the two RS-232 ports as Command parser and DPM4000 Interface is not possible.



If the Error check of connected device configuration is selected for one of the RS-232 ports, the user can use the corresponding Configure button to open the Error Check Parameters window to set interface transmission method and polling parameters.



If the user selects the configuration DPM 4000 Interface for one of the RS-232 ports, the user can use the corresponding Configure button to open the DPM 4000 Parameters window. Connecting the N8000 to a ProMatrix/ ProAnnounce DPM 4000 allows setting nodes of a N8000's paging matrix from the DPM 4000. Defining call inputs is limited to a maximum of four call inputs per N8000.



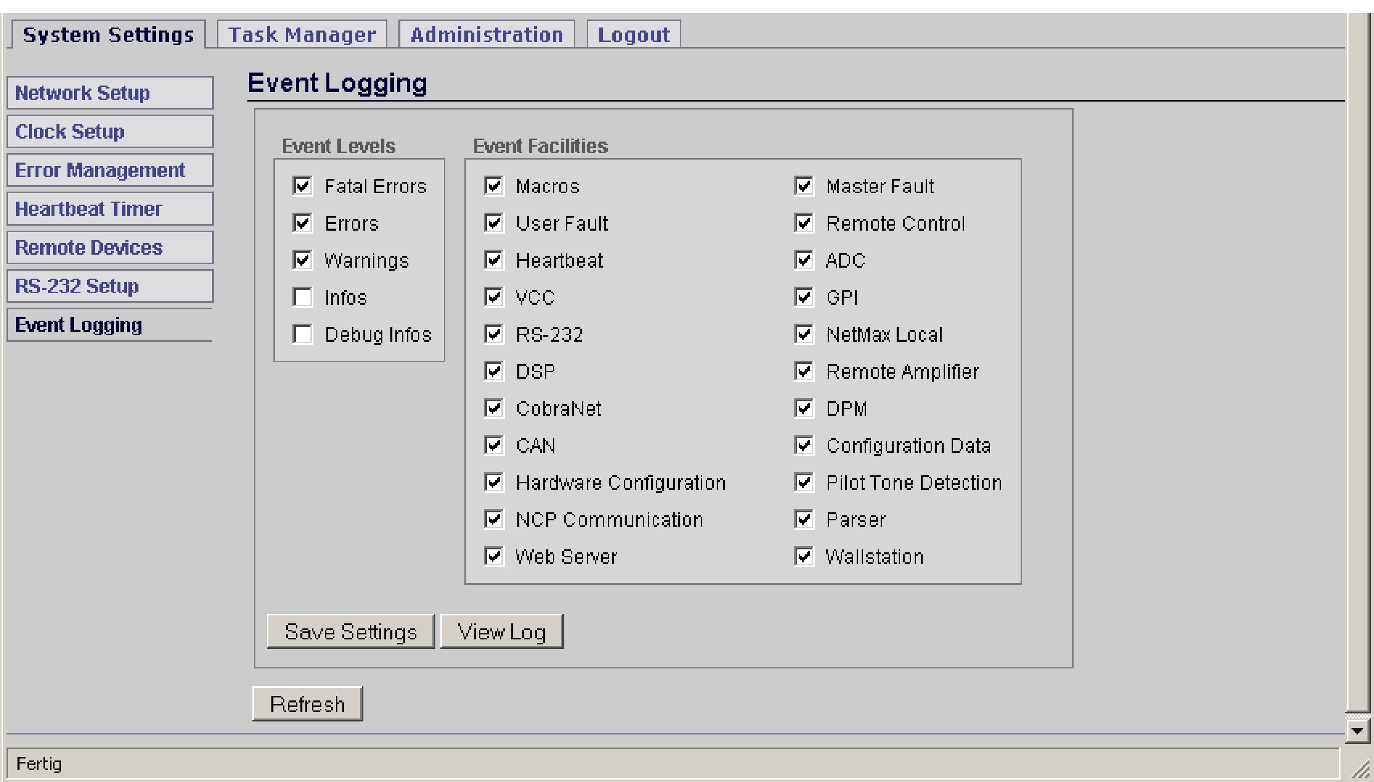
Element	Description
Zone Offset:	Zone Offset specifies the first DPM 4000 calling zone in which the paging matrix begins in the respective matrix row. The entered value has to be in the range of 1 and 100 because the maximum number of possible calling zones is 100.
Zone Width:	Zone Width defines how many consecutive nodes of the corresponding matrix row a DPM 4000 can access. DPM 4000 calling zones to be served by a paging matrix need to be numbered in consecutive order. The entered value has to be in the range of 1 and 32 because the maximum number of nodes included in a row of a priority matrix is 32.

Object ID:	The Object ID specifies the DSP object of the N8000 to be accessed. It is identical for all call inputs of a single paging matrix. However, when using a variety of matrixes, the Object ID has to be different for each matrix. Find the Object ID via Modify Properties --> DSP. Priority Matrix_x.Connect Paging Line OID of the DSP block Priority Matrix.
Matrix Input Line: 1	This parameter defines the paging matrix input, i.e. the matrix row, which will be used as call input. The entered value has to be in the range of 1 and 32 because a priority matrix can include a maximum of 32 rows.
Matrix Rows: 2	Enter the number of rows of the priority matrix.
Matrix Columns: 2	Enter the number of columns of the priority matrix.
Signal: Alarm, Chime, Message, Paging1, Paging2, Paging3, Paging4, BProg	This parameter defines which types of audio signals (DPM 4000 internal and external) will trigger a specific input of the paging matrix.

Event Logging

The Event Logging page allows specifying which types of events will be written to the Log File.

– Problems (errors) that occurred in the NetMax system and
– System status or change in state messages are referred to as events. Event types can be selected in two different ways: by event level or by event facility, where the latter describes a specific system part of the N8000.



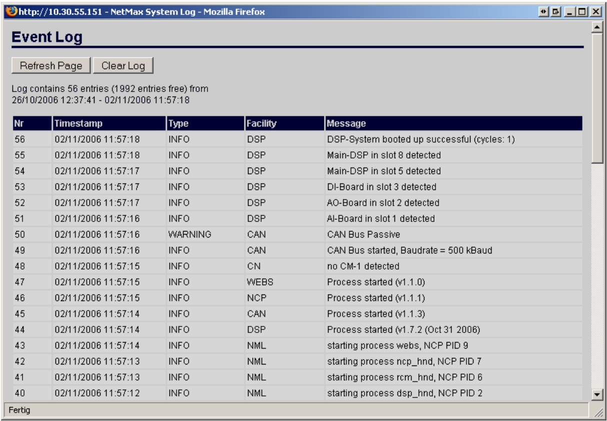
Event Levels

Element	Description
Fatal Errors	Severe operating system errors which can lead to a system crash
Errors	Errors on the application level
Warnings	Warnings
Infos	Information
Debug Infos	Detailed debugging information

Event Facilities

Element	Description
Macros	Information about the Task Manager making a Macro-Call
User Fault	Information about changing the Error Code of User Faults
Heartbeat	Information concerning the Heartbeat Timer
VCC	Information concerning internal Supply Voltages
RS-232	Information concerning the RS-232 port
DSP	DSP system information
CobraNet	CobraNet system information
CAN	CAN interface information
Hardware Configuration	Information about the Hardware Configuration
NCP Communication	NCP protocol information
Web Server	Web Server information
Master Fault	Information about changing the Master Fault
Remote Control	Information concerning the RCM
ADC	Information concerning the A/D converter
GPI	Information concerning the Control Port (General Purpose Interface)
NetMax Local	Information about local events within the N8000
Remote Amplifier	Information concerning the monitoring of Remote Amps
DPM	ProMatrix application information
Configuration Data	Configuration Data information
Pilot Tone Detection	Pilot Tone Detection information
Parser	ASCII Control Protocol information
Wall station	Information concerning connected Wall stations

The Event Log window lists the events in consecutive order including date and time stamp.



http://10.30.55.151 - NetMax System Log - Mozilla Firefox

Event Log

Refresh Page Clear Log

Log contains 56 entries (1992 entries free) from
26/10/2006 12:37:41 - 02/11/2006 11:57:18

Nr	Timestamp	Type	Facility	Message
56	02/11/2006 11:57:18	INFO	DSP	DSP-System booted up successful (cycles: 1)
55	02/11/2006 11:57:18	INFO	DSP	Main-DSP in slot 8 detected
54	02/11/2006 11:57:17	INFO	DSP	Main-DSP in slot 5 detected
53	02/11/2006 11:57:17	INFO	DSP	DI-Board in slot 3 detected
52	02/11/2006 11:57:17	INFO	DSP	AO-Board in slot 2 detected
51	02/11/2006 11:57:16	INFO	DSP	AI-Board in slot 1 detected
50	02/11/2006 11:57:16	WARNING	CAN	CAN Bus Passive
49	02/11/2006 11:57:16	INFO	CAN	CAN Bus started, Baudrate = 500 kBaud
48	02/11/2006 11:57:15	INFO	CN	no CM-1 detected
47	02/11/2006 11:57:15	INFO	WEBS	Process started (v1.1.0)
46	02/11/2006 11:57:15	INFO	NCP	Process started (v1.1.1)
45	02/11/2006 11:57:14	INFO	CAN	Process started (v1.1.3)
44	02/11/2006 11:57:14	INFO	DSP	Process started (v1.7.2 (Oct 31 2006)
43	02/11/2006 11:57:14	INFO	NML	starting process webs, NCP PID 9
42	02/11/2006 11:57:13	INFO	NML	starting process ncp_hnd, NCP PID 7
41	02/11/2006 11:57:13	INFO	NML	starting process rcm_hnd, NCP PID 6
40	02/11/2006 11:57:12	INFO	NML	starting process dsp_hnd, NCP PID 2

Fertig

Element	Description
Refresh Page	Rereads the Event Log of the N8000.
Clear Log	The entire Event Log currently existing in the N8000 will be deleted.
Nr	The entries in the Event Log are numbered in ascending order.
Timestamp	Date and time of the event.
Type	Indicates the level of an event. The Event Log only lists events for which an Event Level has been specified on the Event Logging page.
Facility	Indicates the subsystem for the concerned event. The Event Log only includes entries for which a Facility has been specified on the Event Logging page.
Message	Description of an event.

TASK MANAGER

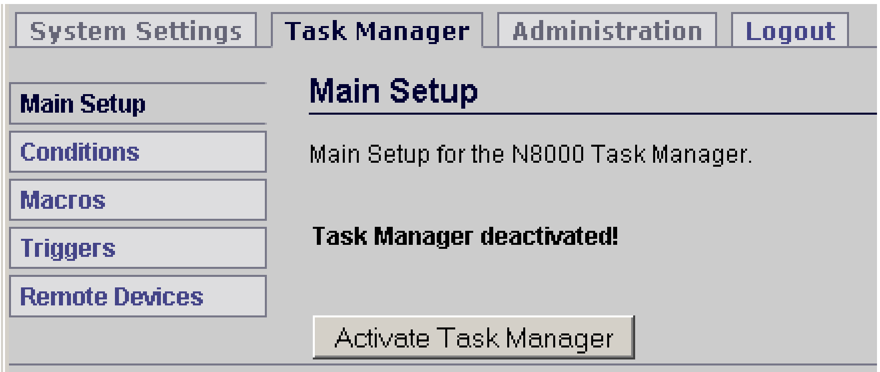
The N8000 Browser Interface is divided into three main windows: System Settings, Task Manager and Administration. Menu entries of the Task Manager window are listed in the following table:

Menu entry	Description
Main Setup	Activating/Deactivating the Task Manager
Conditions	Configuration of up to 100 conditions
Macros	Configuration of up to 100 macros
Triggers	Configuration of up to 100 triggers
Remote Devices	Configuration of up to 100 remote devices

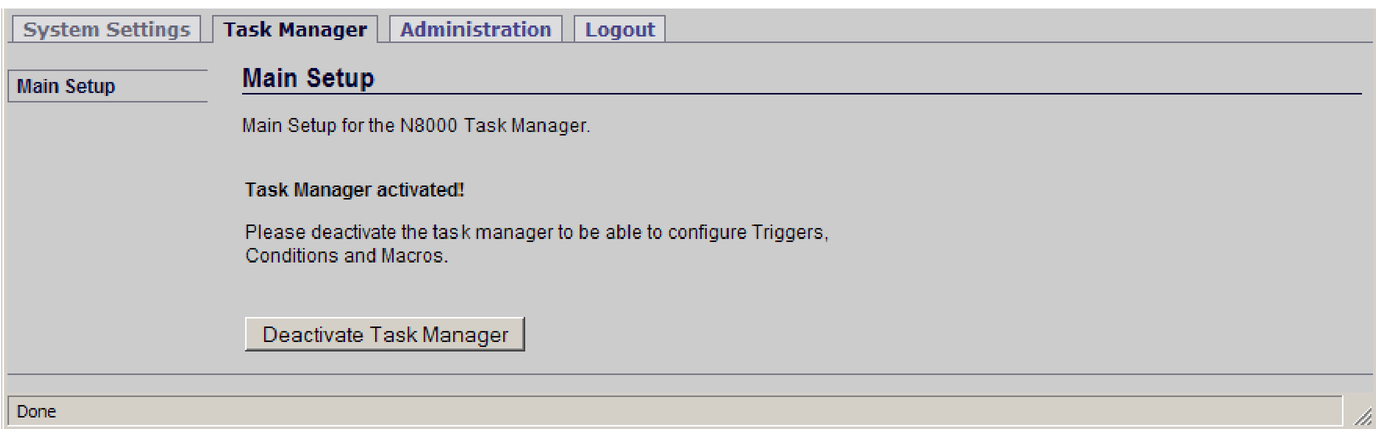
Main Setup

The Main Setup page allows activating or deactivating the Task Manager of the N8000.

HINT: Editing Conditions, Macros and Triggers is only possible when the Task Manager is deactivated.

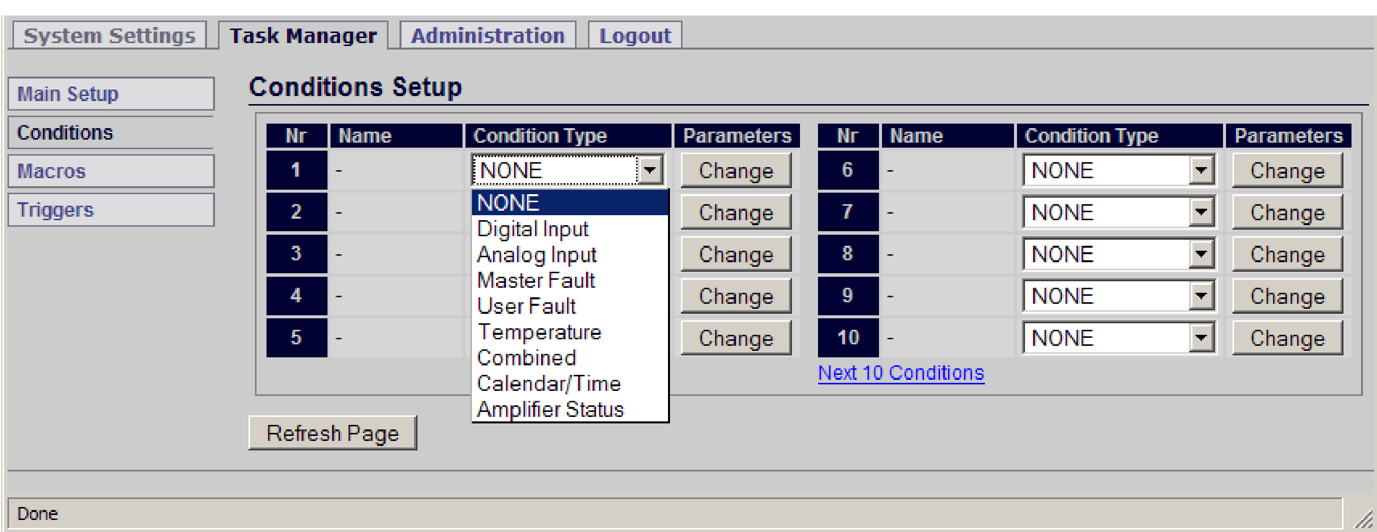


Editing conditions, macros and triggers is not possible with the Task Manager being active because their menus are grayed out and cannot be accessed.



Conditions

The N8000 is capable of managing up to 100 Conditions. Whether commands are being executed depends on Conditions. Commands to be executed are defined as Macros. Whether a Conditions result is true or false depends on the type of Condition and its respective parameters.



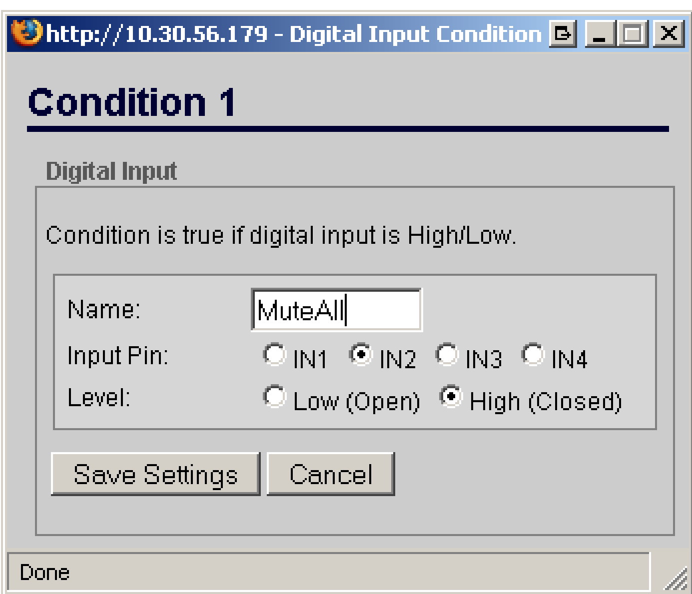
The N8000 provides the following types of conditions:
- Digital Input
- Analog Input
- Master Fault
- User Fault
- Temperature
- Combined
- Calendar/Time
- Amplifier Status

The following section explains each type and the related parameters for each condition. Condition names can be composed of maximally 10 characters.

Digital Input: Using the four control inputs of the Control Port as digital inputs is possible. The status of a control input is Low (Open) when it is not connected to the reference voltage of 10 V. In case it is connected to a reference voltage of 10 V, the status of a control input is High (Closed). 2.5 V marks the threshold between High and Low. The condition has the same status as the selected control input.

Example:

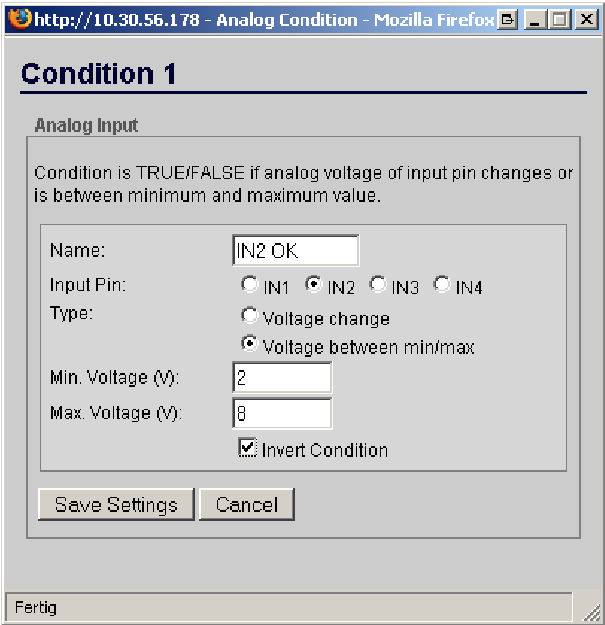
The screenshot on the left represents the condition "Mute All", which monitors a switch of control input 2. The conditions result equals true" as long as the normally open contact is closed, i.e. the control input 2 is connected to 10 volts.



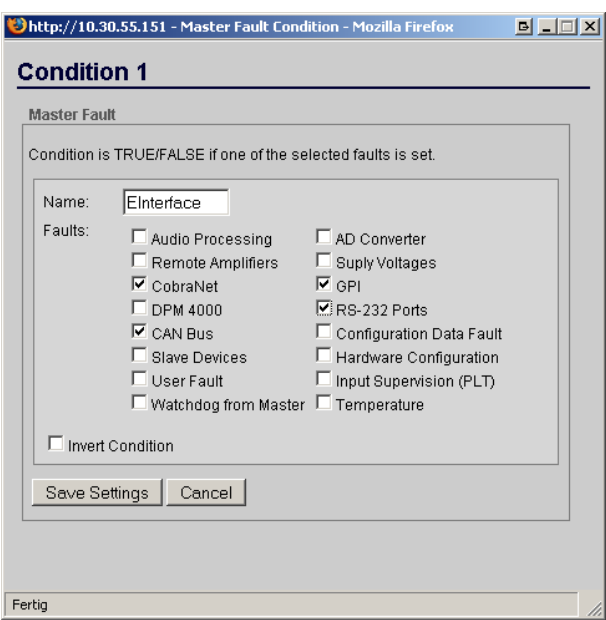
Analog Input: The input voltage at the control input of the Control Port can be measured and the result can be compared with upper and lower thresholds. The condition uses the result of this comparison. The conditions result is true (or false when choosing Invert Condition instead), as long as the input voltage is within the specified voltage range.

Example:

The screenshot on the left shows the condition "IN2_OK" that monitors the voltage level of control input 2. This example represents a case, when a control input is used as digital input. The status is defined in the voltage ranges of 0V...2V and 8V...10V only. On the other hand, if the input voltage is in the range of 2 volts and 8 volts, that is, if the input voltage is in a voltage range that is not feasible for using the inputs as digital input, the condition is true which results in error indication.



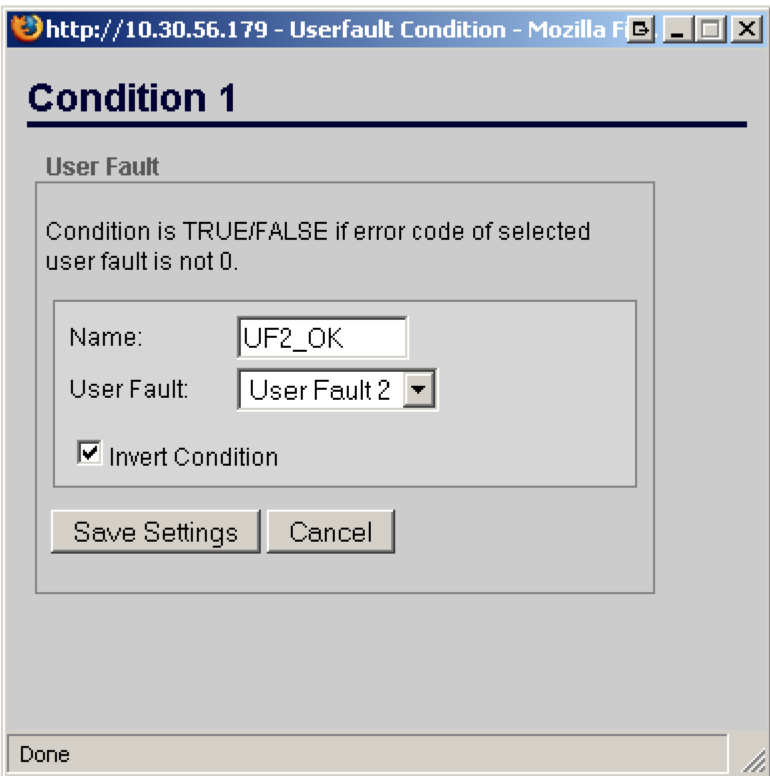
Master Fault: A condition can depend on a selected internal N8000 error. The conditions result is true (or false when choosing Invert Condition instead) if one or various errors of the selected type occur.
Example:
The screenshot on the left shows the condition "EInterface". The conditions result is true whenever a fault is recognized on the N8000 interface. The following types of errors are available to select from: CobraNet, CAN Bus, GPI and RS-232 interface.



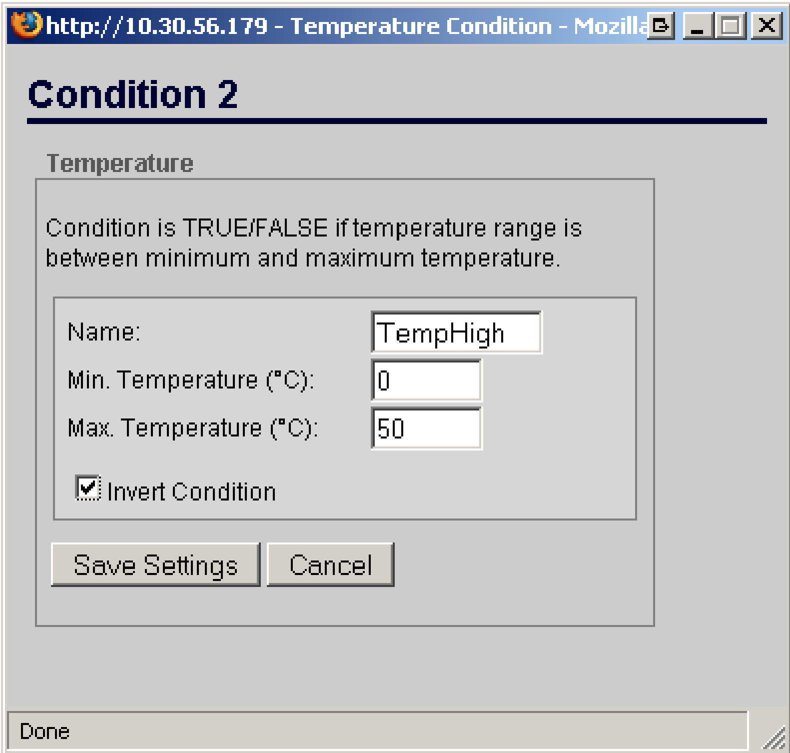
User Fault: It is possible to select a User Fault on which a condition depends. The conditions result is true (or false when choosing Invert Condition instead) once the User Fault occurs.

HINT: A User Fault occurs if the associated Error Code is not equal 0. The User Fault macro is always used when setting a User Fault.

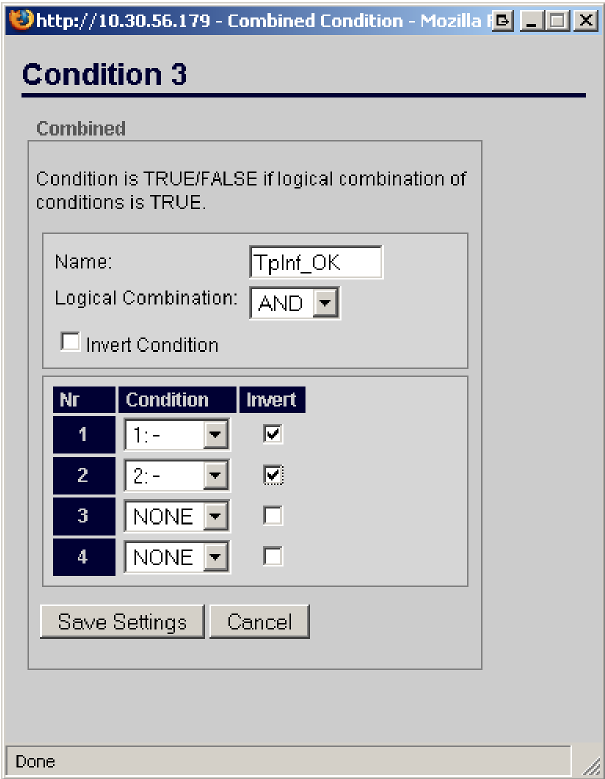
Example:
The screenshot on the left shows the condition "UserFault2OK". This conditions result is true if the User Fault 2 has not been set, i.e. its Error Code equals 0.



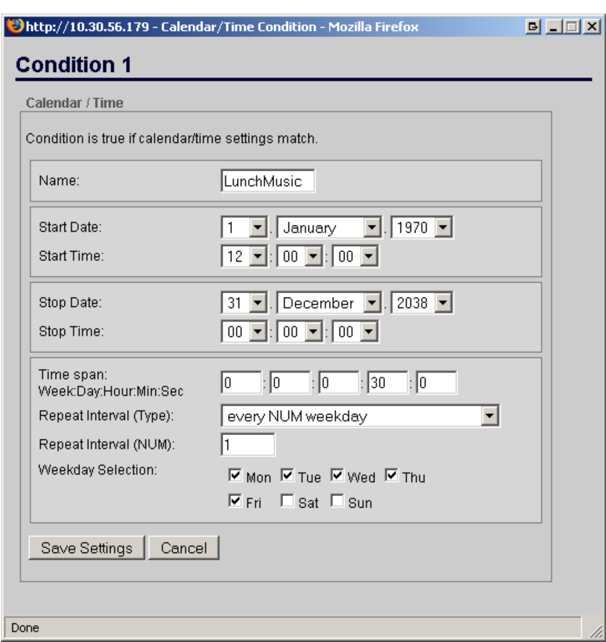
Temperature: The temperature within the N8000 System Controllers enclosure is constantly being monitored. The condition depends on the compliance with upper and lower thermal thresholds. Limit values have to be entered in °C. The conditions result is true (or false when choosing Invert Condition instead) as long as the actual temperature stays within the defined temperature range.
HINT: Converting temperature values from degrees Fahrenheit °F to degrees Celsius °C is done according to the following formula: degree Celsius = (degree Fahrenheit - 32) * 5/9
Example:
The screenshot on the left shows the condition "Temp High". This conditions result is true as long as the temperature stays above the maximum allowable temperature of 50°C.



Combined: This condition allows logically combining up to four other conditions. Logical operators that can be used are AND, OR and XOR. The conditions result is true (or false when choosing Invert Condition instead) if the logic operation of the selected condition is true.
Example:
The screenshot on the left shows the condition "TpInf_OK". This conditions result is true as long as the temperature has not exceeded the upper limit value and at the same time no error has occurred at the interfaces of the N8000. This is achieved by inverting Condition 1 "TempHigh" and Condition 2 "EInterface" and combining the results with the logic operation AND.

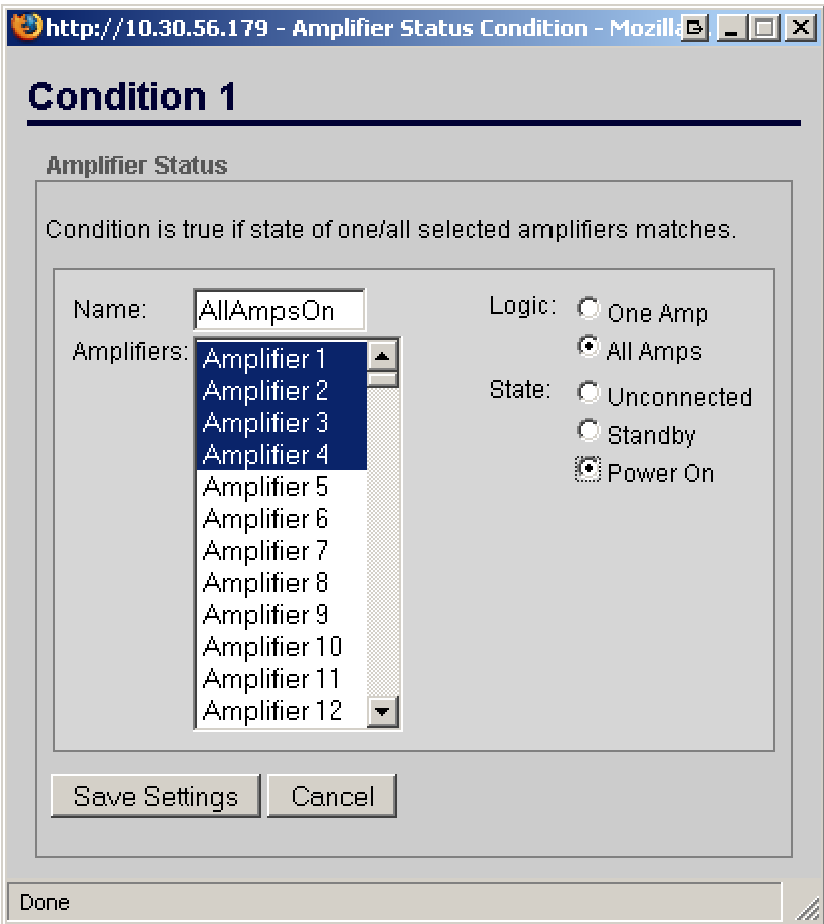


Calendar/Time: The condition compares the current time (according to the system clock) with user-definable non-recurring or periodically recurring time spans. The conditions result is true within the defined time span(s). Example: The screenshot on the left shows the condition "LunchMusic". This conditions result is true for 30 minutes on each day at noon (except for Saturdays and Sundays).

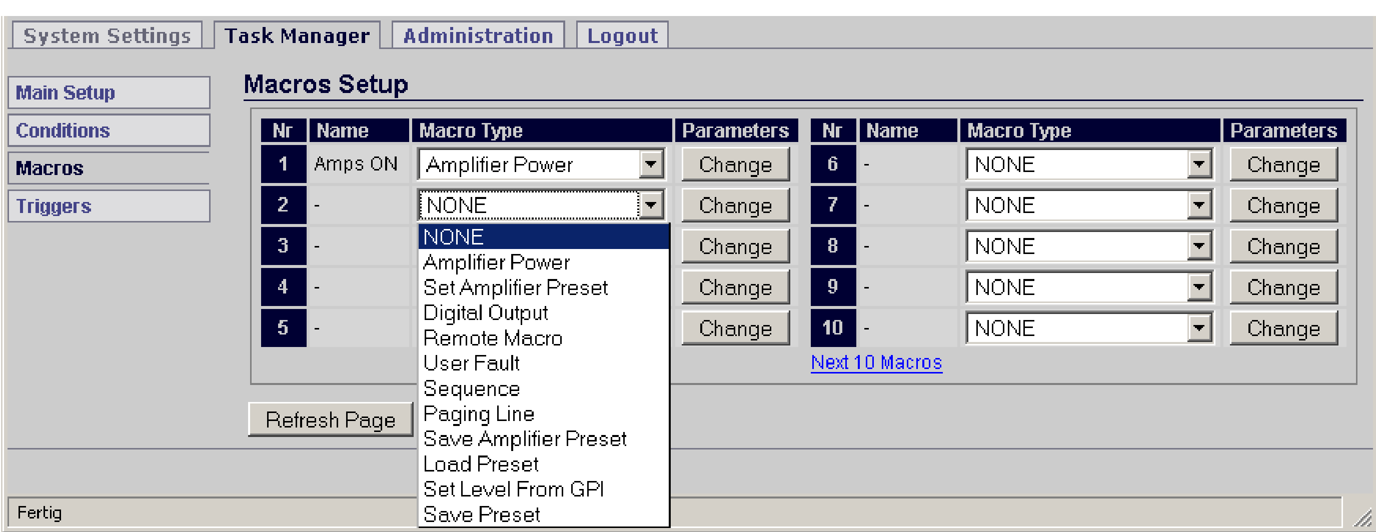


Amplifier Status: This condition depends on the status of a (or the identical states of several) power amp(s). Amps specified in the amplifier list with "Amplifier X" represent remote amps with the CAN address X that are connected to the CAN bus. The conditions result is true if one or all of the selected amplifiers are in the selected status.

Example:
The screenshot on the left shows the condition "All Amps On". This conditions result is true whenever all power amps with CAN addresses between 1 and 4 are powered on.

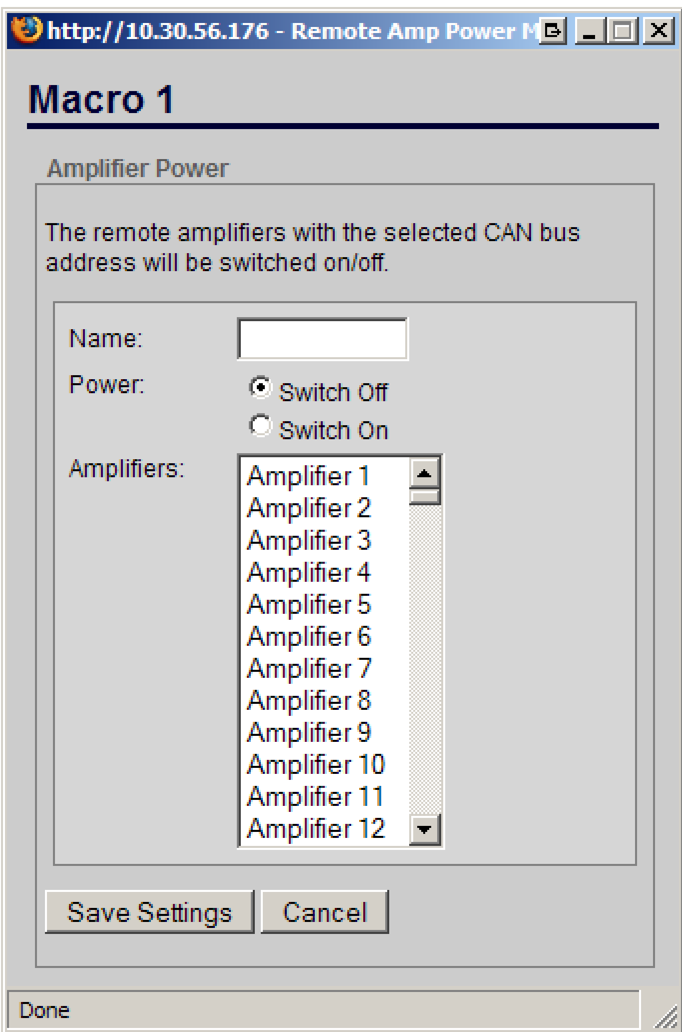


Macros
The N8000 is capable of managing up to 100 macros. Macros are used to change the state of the N8000 or the states of connected remote amps.

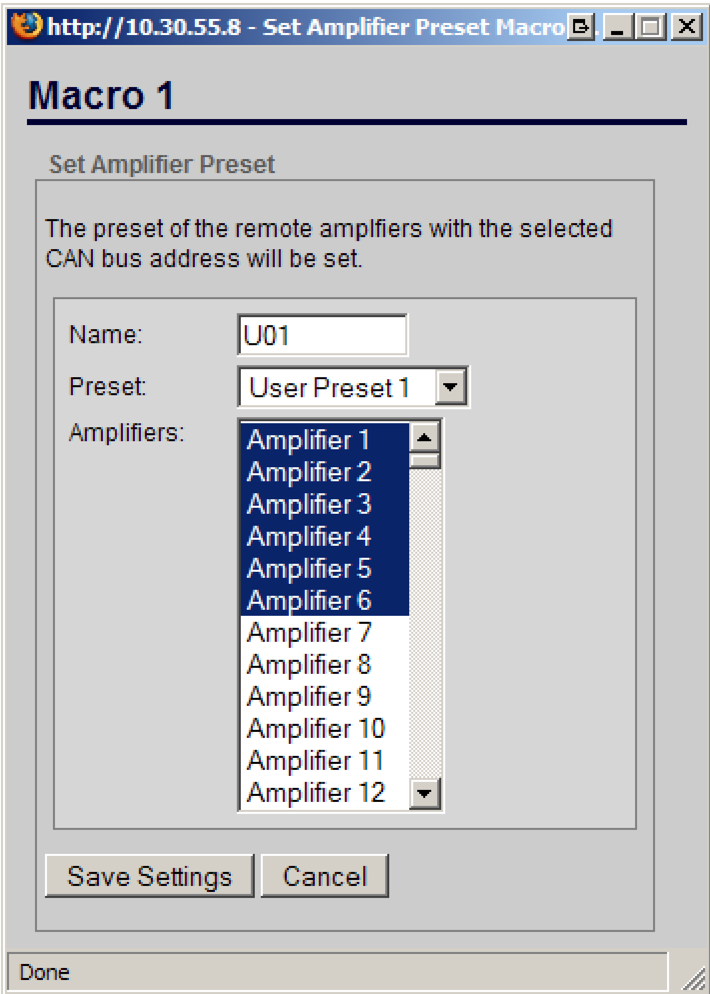


The N8000 provides the following macro types:
- Amplifier Power
- Set Amplifier Preset
- Digital Output
- Remote Macro
- User Fault
- Sequence
- Paging Line
- Save Amplifier Preset
- Load Preset
- Set Level From GPI
- Save N8000 Preset

Amplifier Power: Execution of this macro switches power amps on or off. Amps specified in the amplifier list as "Amplifier X" represent remote amps with CAN address ”X” that are connected to the CAN bus.

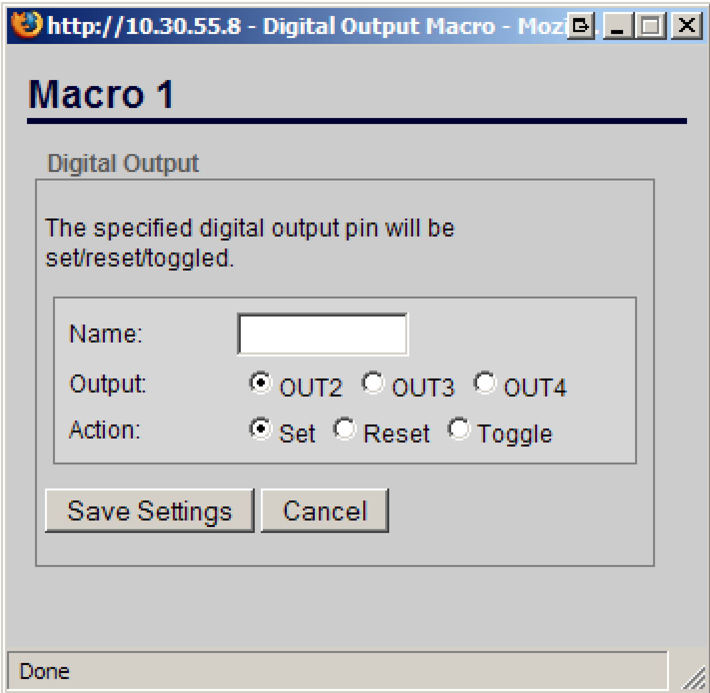


Set Amplifier Preset: Execution of this macro loads the selected Preset into memory of a single or a variety of power amps. Amps specified in the amplifier list as „Amplifier X“ represent remote amps with CAN address „X“ that are connected to the CAN bus.

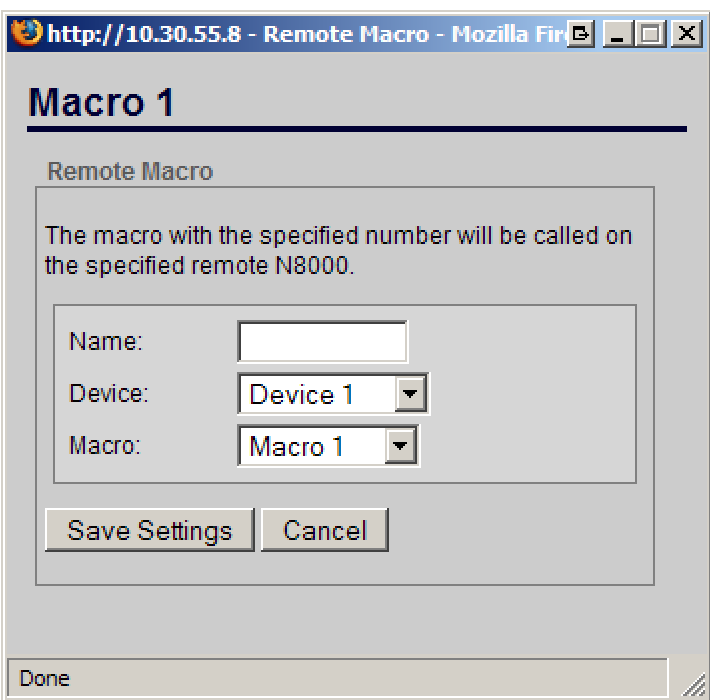


Digital Output: Execution of this macro changes the (digital) state of a control output of the N8000. It is possible to

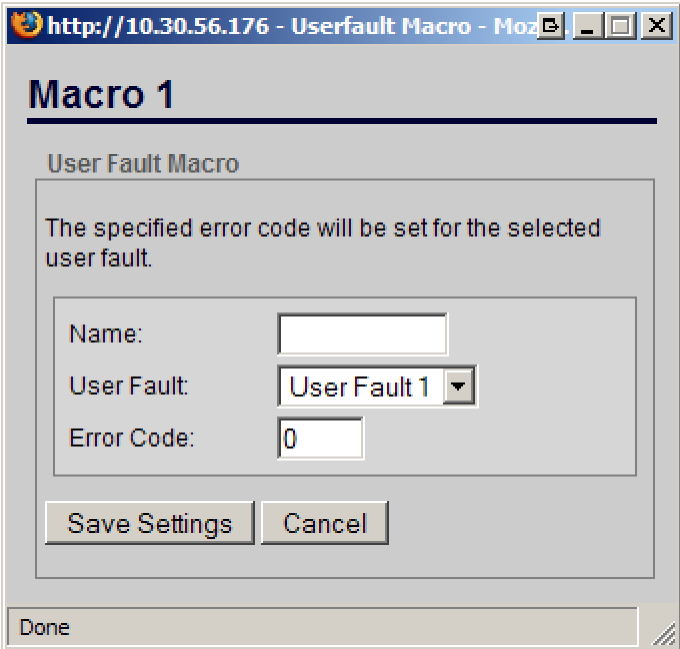
– set (relay contact closed, i.e. connected to ground),
– reset (relay contact open) or
– invert a control output.



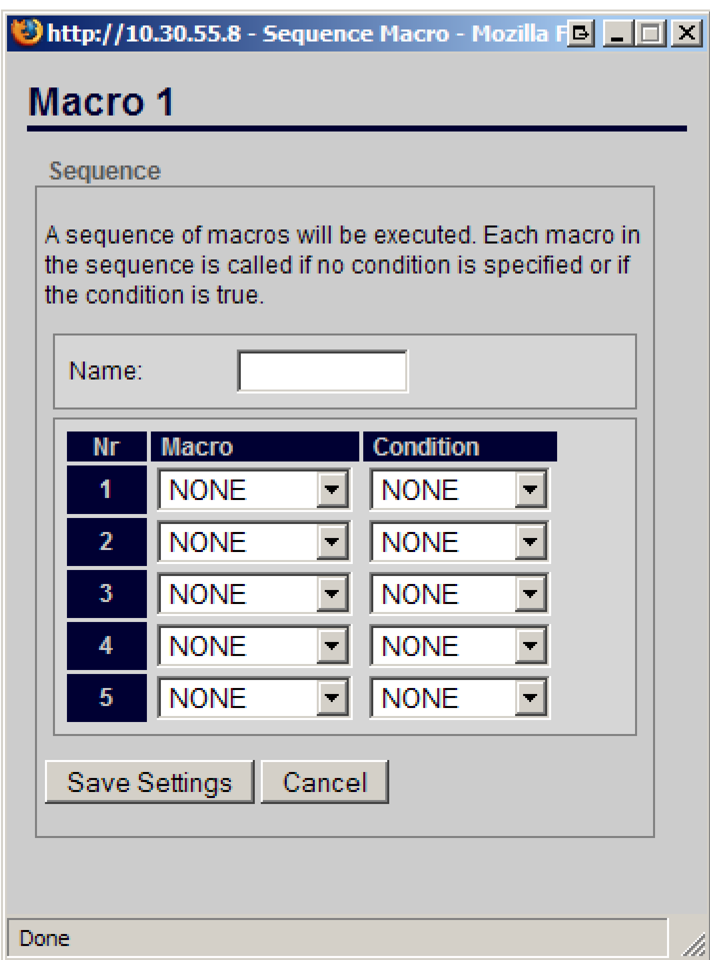
Remote Macro: Execution of this macro executes a freely selectable macro on another N8000. Defining the N8000s that appear in the Device list has to be done in the Remote Devices window.



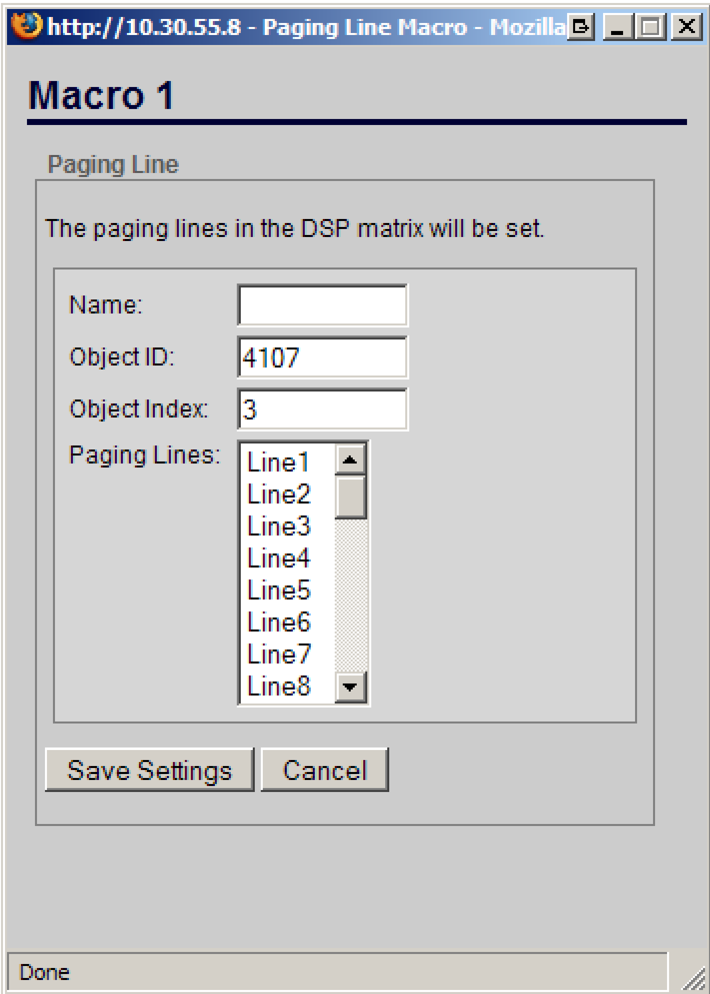
User Fault: Execution of this macro assigns a random numerical Error Code to the selected User Fault..
HINT: ErrorCode"0"represents"noerror".



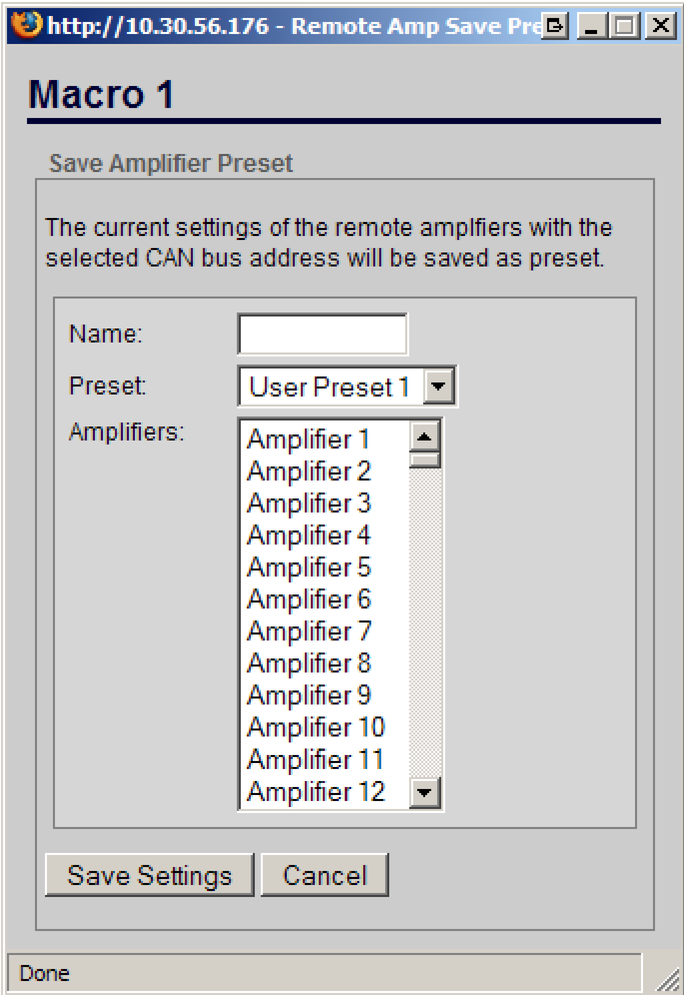
Sequence: Execution of this macro sequentially executes up to five other macros. Each macro of a sequence can depend on a condition. Conditions are defined on the Conditions page.



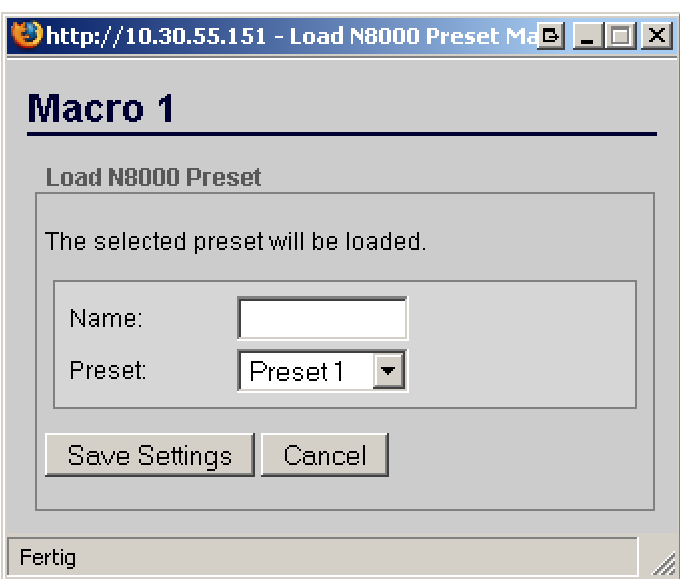
Paging Line: Connecting the N8000 to a ProMatrix/ProAnnounce DPM 4000 allows setting nodes of a N8000's paging matrix from the DPM4000. Therefore, it is necessary to select the macro's Parameters Object ID, Object Index and Paging Lines.



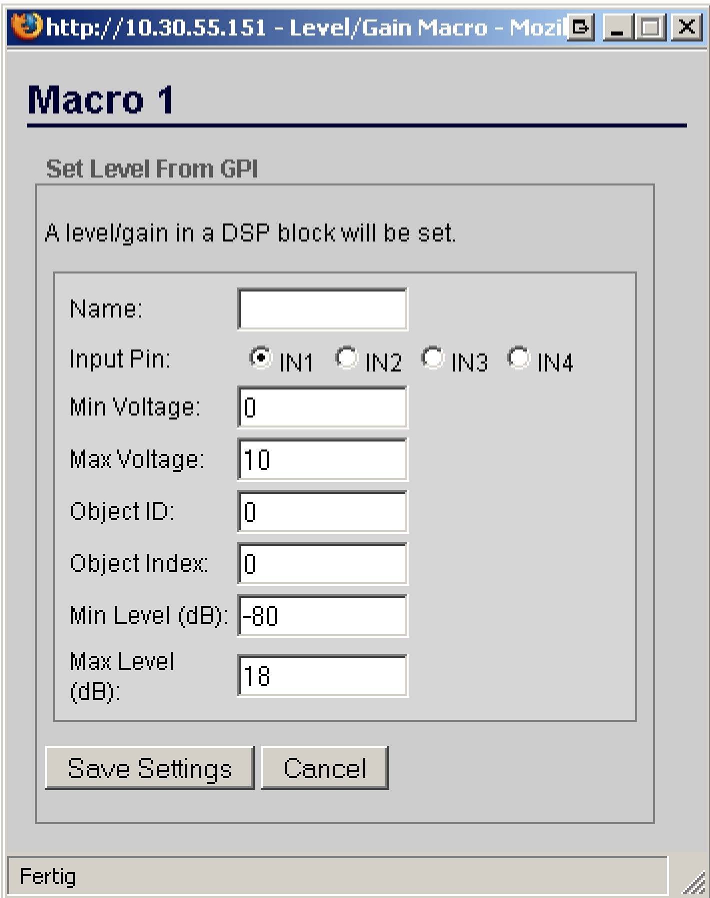
Save Amplifier Preset: The current settings of the selected amps are stored in a selected User Preset. Amps specified in the amplifier list as „Amplifier X" represent remote amps with CAN address „X" that are connected to the CAN bus.



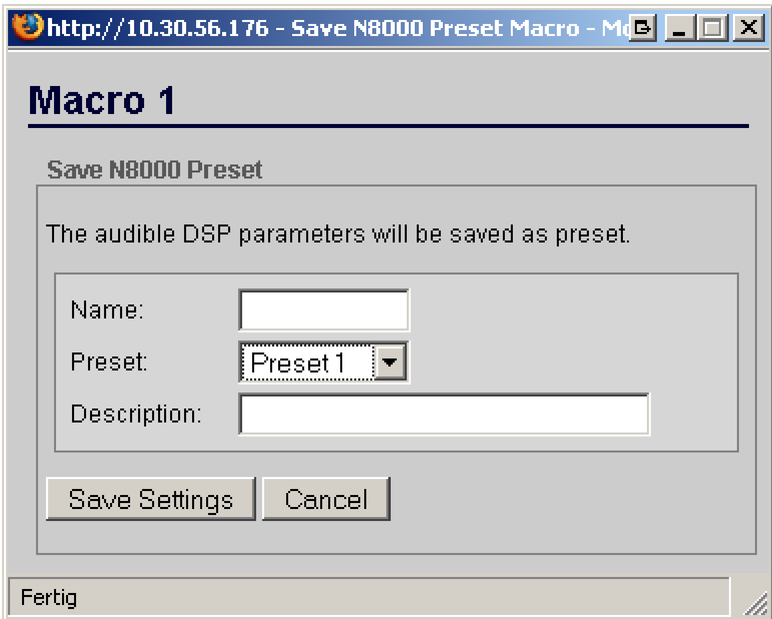
LoadN8000Preset. Execution of this macro loads a selected Preset into the N8000.



Set Level From GPI: This macro allows using a control voltage with definable value range for the level control of a N8000 DSP block. Therefore, it is necessary to select the DSP block's parameters Object ID and Object Index. The Min Voltage and. Max Voltage fields are for defining an allowable value range of the voltage present at the selected input. The Min Level (dB) and Max Level (dB) fields define the value range for the absolute level, which, in relation to the applied voltage, is sent to an object.

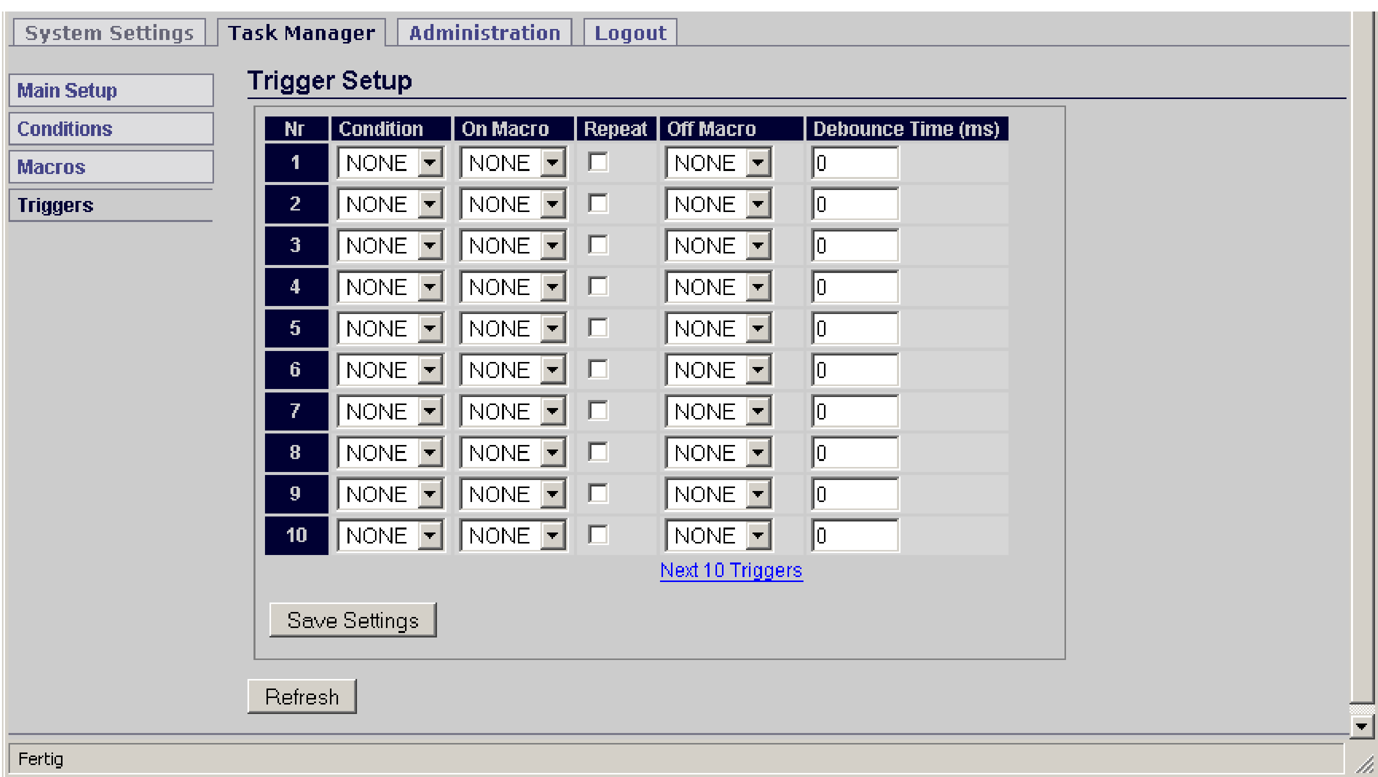


Save N8000 Preset: The macro saves the current settings of the N8000's DSP configuration in a Preset. Preset number and preset name are user-definable.



Triggers

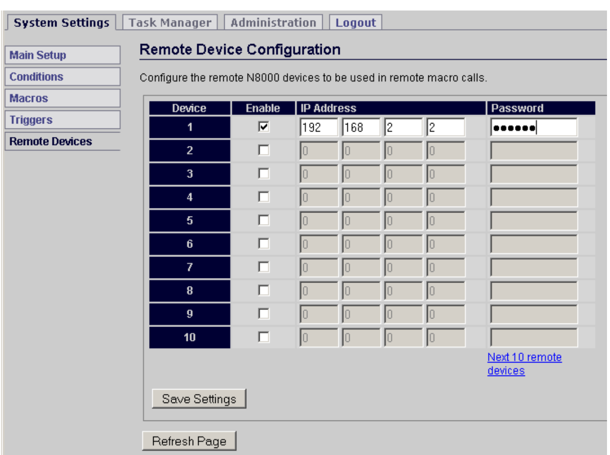
Triggers control the execution of macros depending on a specific state of a Condition. Up to 100 triggers can be employed.



Element	Description
Nr	Triggers are numbered in ascending order. It is possible to manage up to 100 triggers.
Condition	The status change of this Condition “triggers” the execution of macros.
On Macro	The selected macro is executed exactly one time when the condition’s state changes from “false” to “ true”.
Repeat	Choosing this option repeats the On macro for the Debounce Time period as long as the condition’s result is „true“. For example, this becomes necessary when the Set Level From GPI macro is triggered by the Analog Input condition.
Off Macro	The selected macro is executed exactly one time when the condition’s state changes from “true” to “false”.
Debounce Time (ms)	It is possible to stipulate a very short time interval during which the status change of a condition needs to be maintained. Execution of the macro is discarded if the status changes back before the Debounce Time has elapsed.

Remote Devices

The Remote Devices page allows configuring the remote N8000 devices used in remote macro calls.



Element	Description
Device	Up to 100 remote N8000 can be used in remote macros.
Enable	Allows deactivating a remote N8000 temporarily.
IP Address	The IP address of the remote N8000.
Password	The password of the user account „netmax“ of the remote N8000.

ADMINISTRATION

The browser interface is divided into three main windows: System Settings, Task Manager and Administration. The following table lists the menu entries in the Administration window:

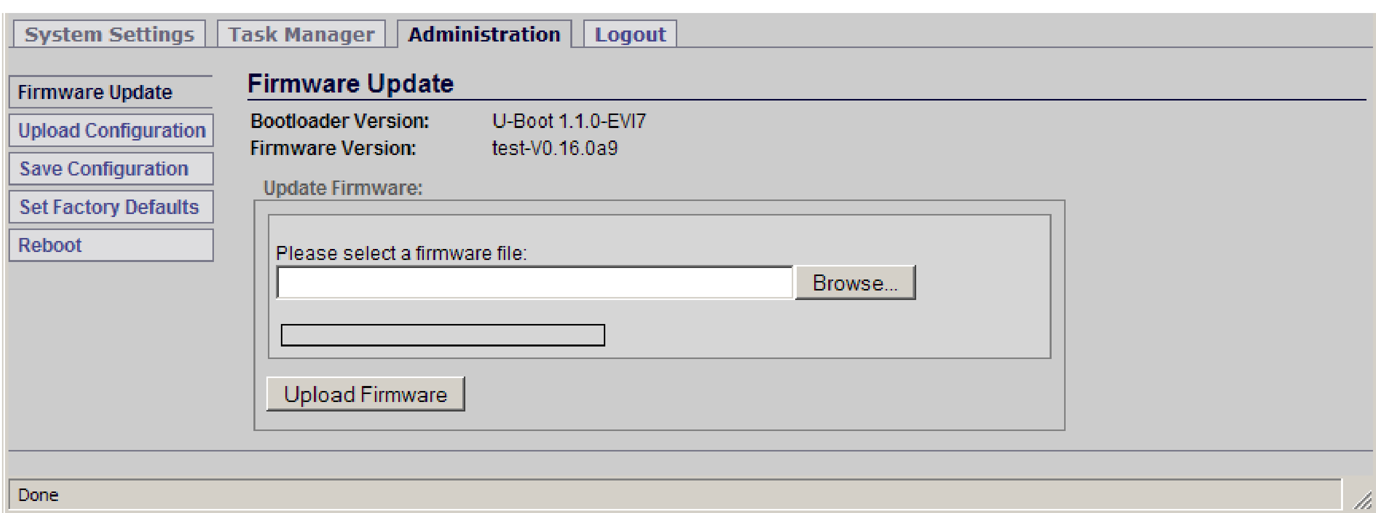
Menu entry	Description
Firmware Update	Updates the firmware of the N8000.
Upload Configuration	Replaces the N8000’s current configuration with a previously saved configuration file.
Save Configuration	Stores the N8000’s current configuration in a configuration file on a PC.
Set Factory Defaults	Resets all parameters of the N8000 to their factory default values.
Reboot	Restarts the N8000.
Set Password	Allows the currently logged-in user to change their password.

Firmware Update

The Firmware Update page provides a convenient mechanism for updating the N8000 firmware.

Caution!
The N8000 firmware should be updated only, if problems with the firmware used so far exist and these can be fixed by using a new firmware version.
Consequences

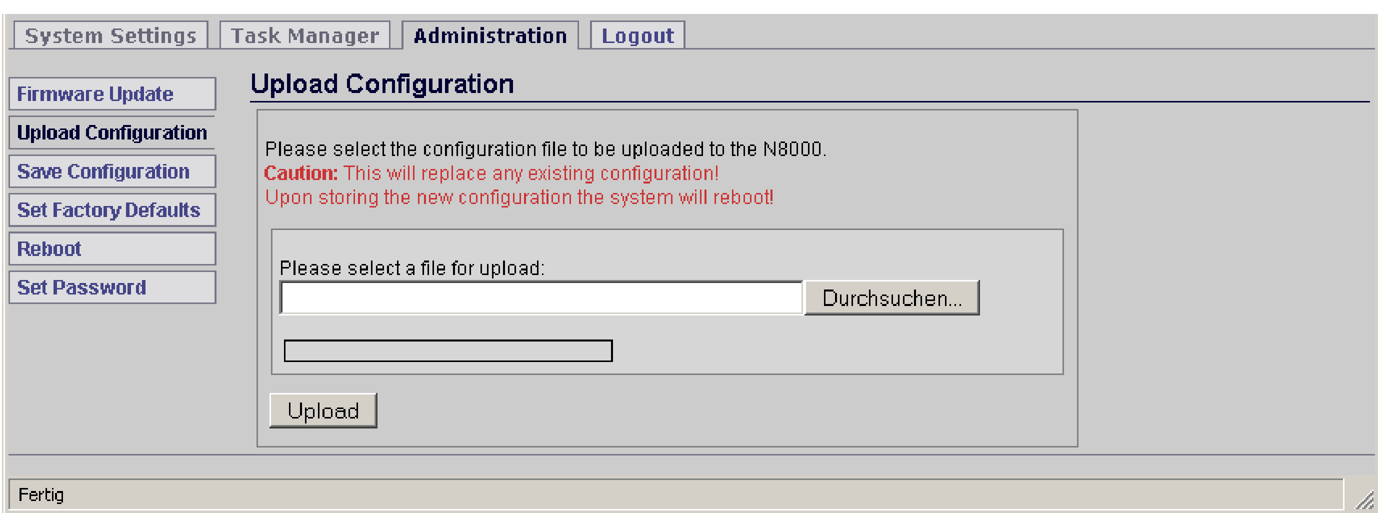
The Browse... button allows the user to navigate through hard drives or storage media (e.g. a CD-ROM) for the appropriate for firmware files. Pressing the Upload Firmware button loads the selected firmware file into the memory buffer of the N8000.



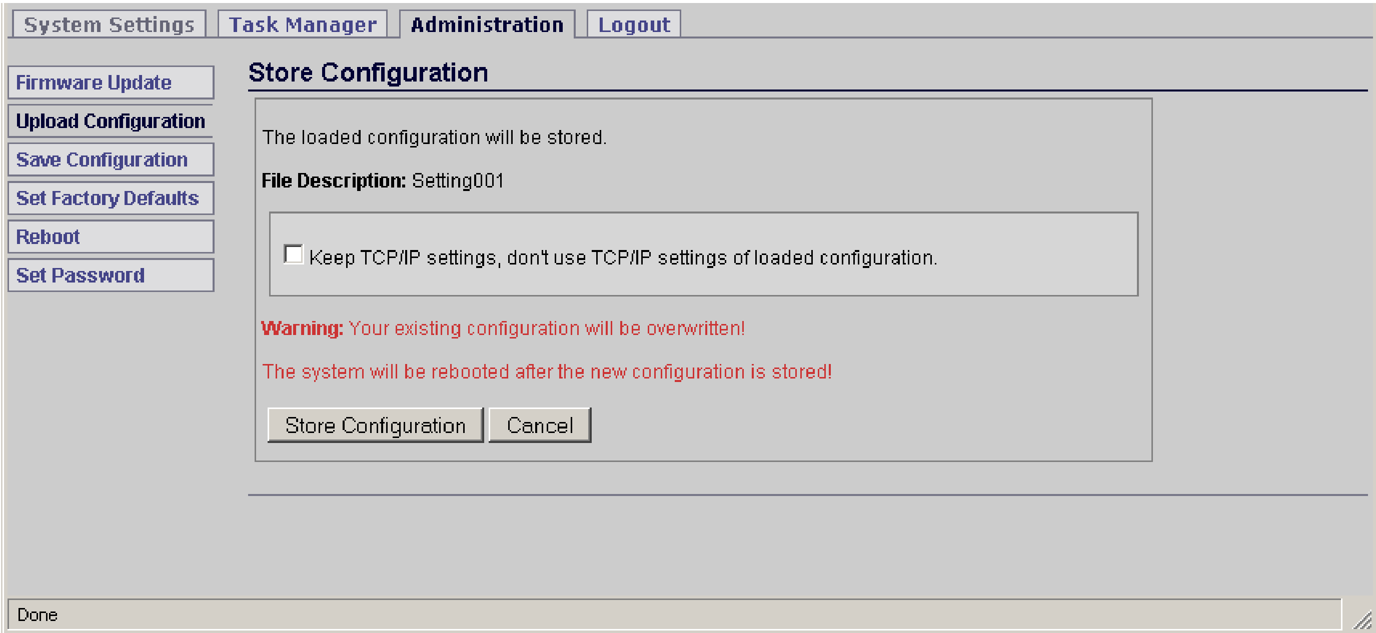
Pressing the Update button writes the new firmware into the N8000, which automatically reboots upon successfully completing the update process.

Upload Configuration

The Upload Configuration page allows transferring configuration data (stored in a backup file via Save Configuration) back to the N8000. The Browse... button allows searching through any available data storage medium for configuration files. Hitting the Upload button loads the previously selected configuration file into the memory buffer of the N8000.

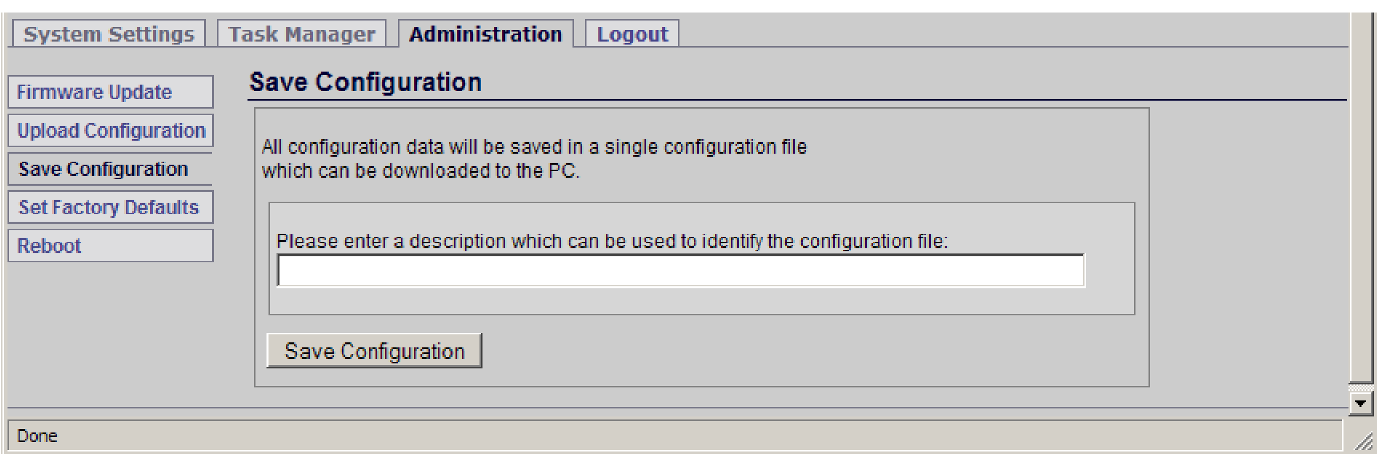


The selected backup file also contains information about the Ethernet port's network configuration. Consequently, storing the newly loaded configuration data into memory of the N8000 also affects the current network configuration. If this is not desired, maintaining the current network configuration is possible by ticking the Keep existing TCP/IP settings... checkbox. Pressing the Store Configuration button actually stores the uploaded configuration data in the N8000.



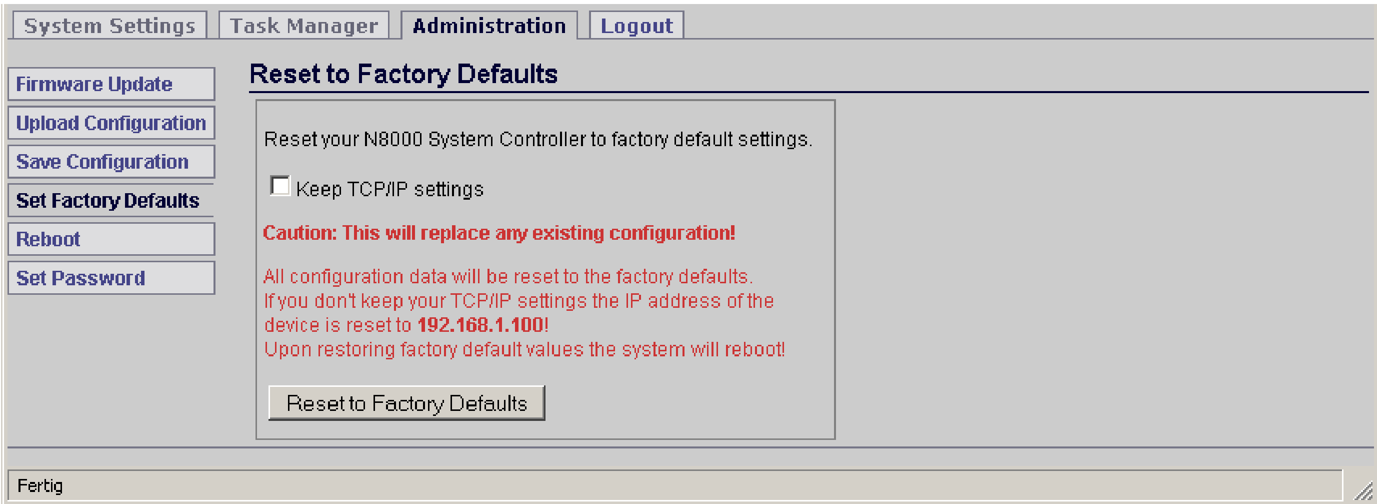
Save Configuration

It is possible to transfer the N8000's entire configuration to the PC and store it in a configuration backup file. This provides an easy option to backup an entire N8000. In addition, loading a backup file (via Upload Configuration) into a number of N8000 System Controllers offers a quick and very convenient way to configure several N8000 absolutely identically.



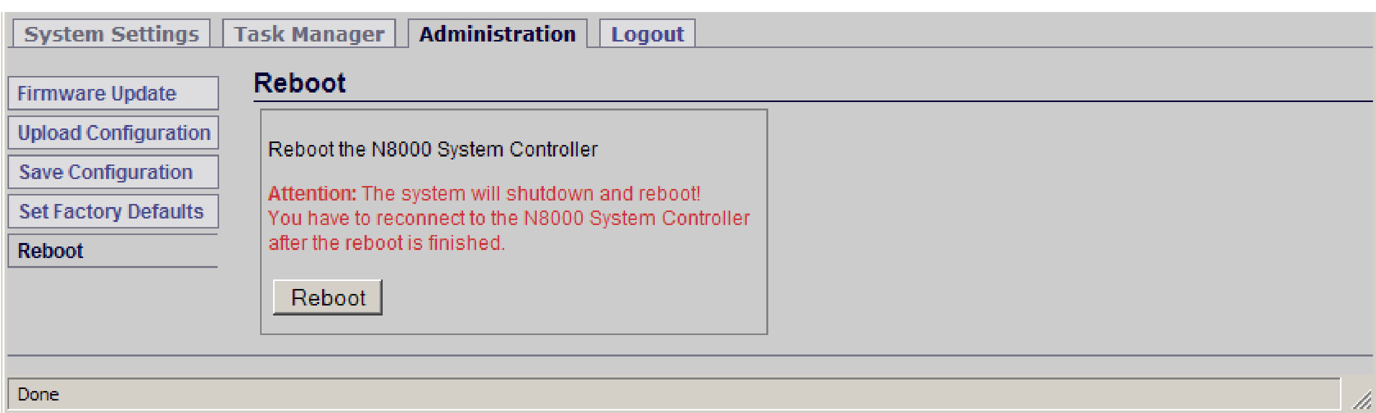
Set Factory Defaults

The Set Factory Defaults page allows the resetting of the N8000 to its factory defaults. The Keep TCP/IP settings option allows the N8000 to be reset to its factory defaults while retaining its network configuration in a situation where the N8000 is currently part of an existing Ethernet network and settings such as IP address must be retained.



Reboot

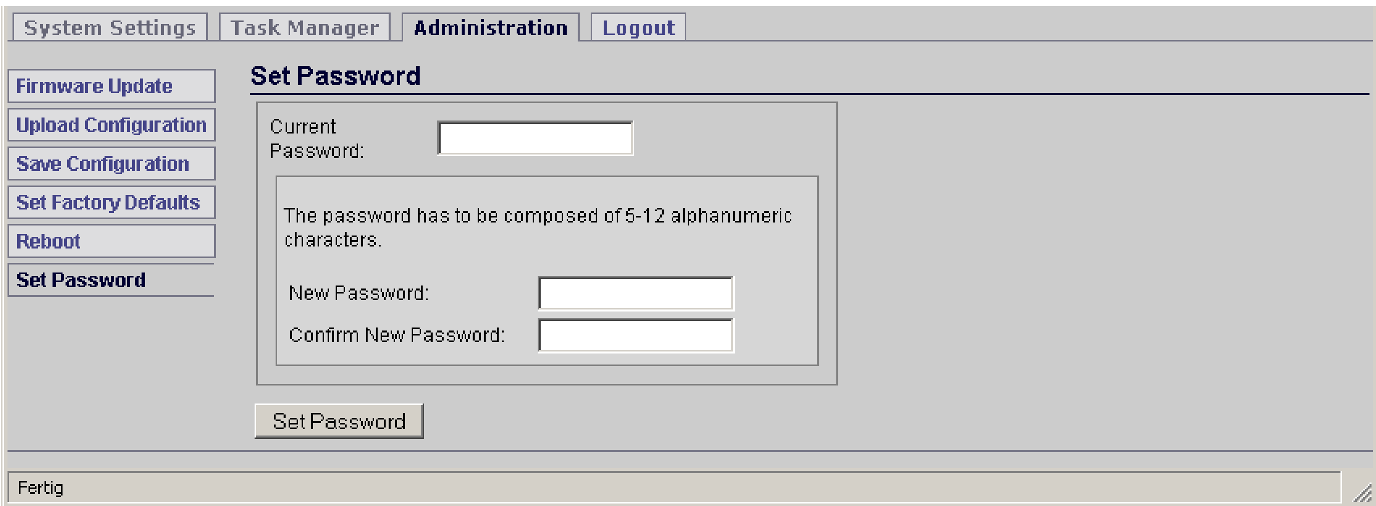
The Reboot page allows rebooting the N8000. Rebooting the N8000 also terminates the current session of the browser interface.



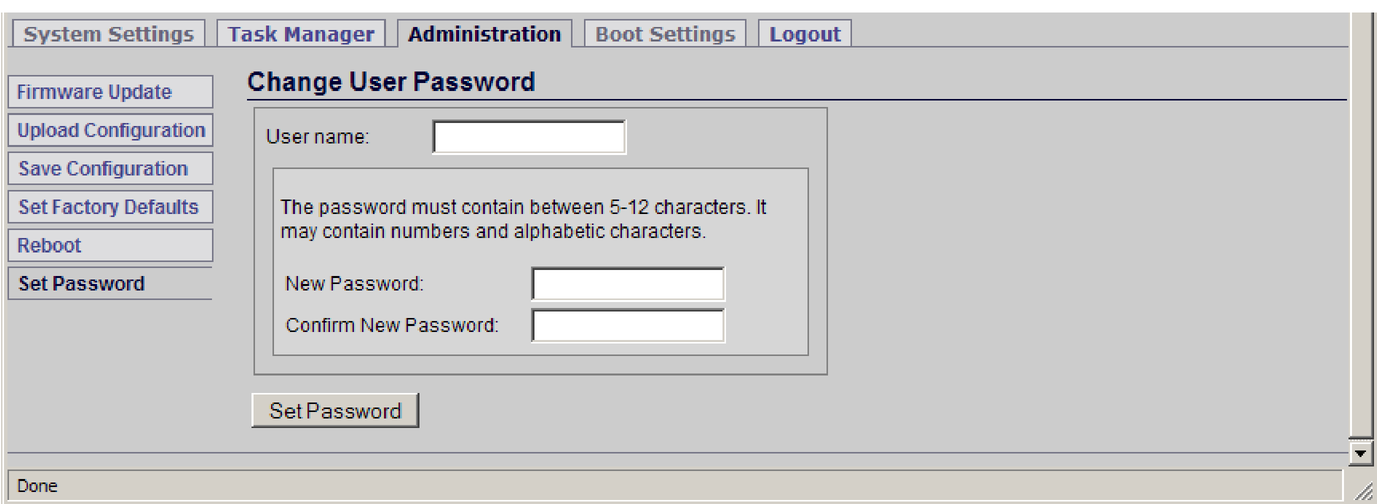
Set Password

The Set Password page allows specifying a new password for the present user of the N8000 browser interface.

HINT: The password has to be composed of at least 5 and up to a maximum of 12 alphanumeric characters; i.e. only alphabetic characters and symbols. The use of special characters is not permissible. The password is case-sensitive.

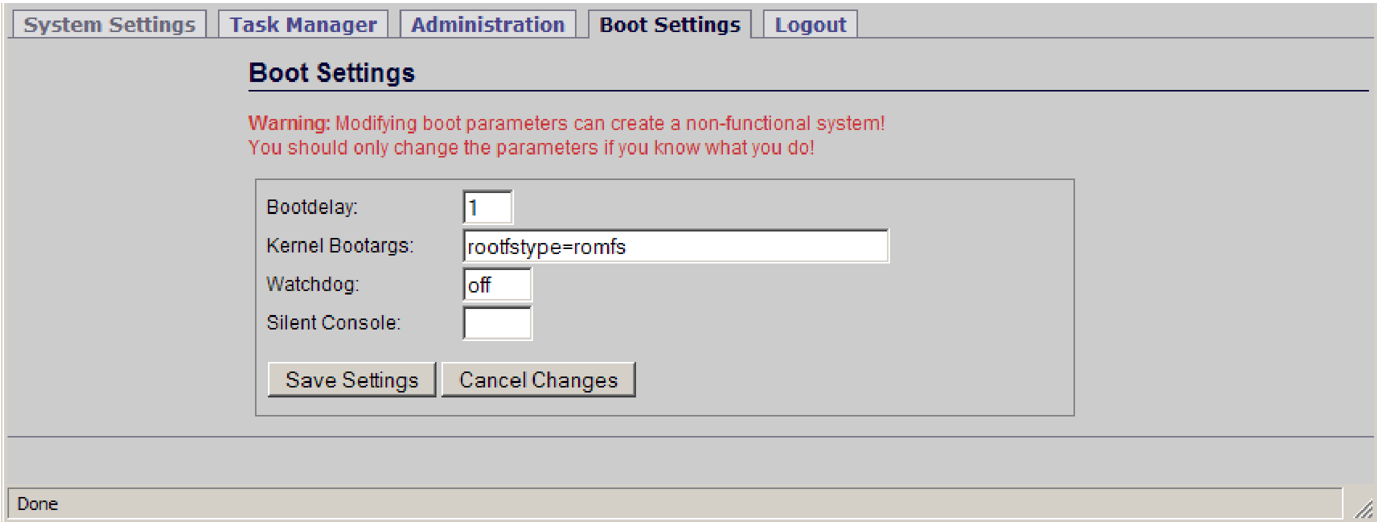


When logged in as administrator, the Change User Password page allows the assignment of new passwords to all users.



BOOT SETTINGS

The Boot Settings page is only accessible if you are logged in as administrator. Changing parameters on this page is not recommended. If you are in doubt, please, leave the pre-set values as they are.

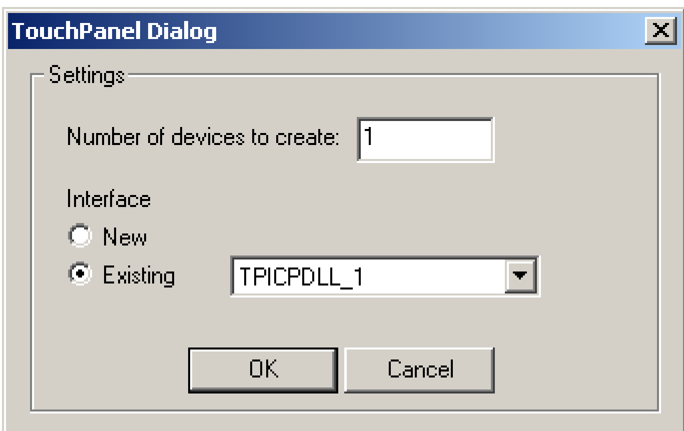


TOUCH PANELS

TPI-5

TPI-5 Device

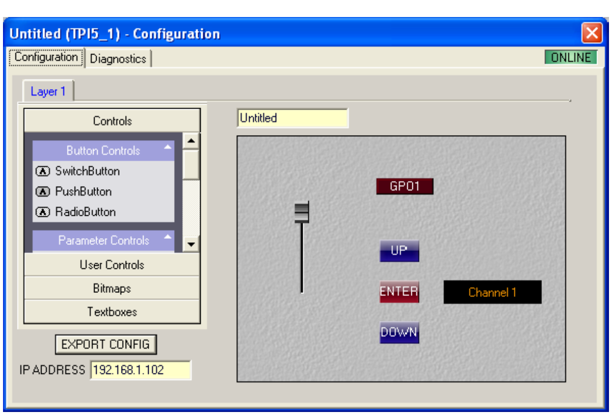
First, create a TPI-5 Device in your IRIS-Net project, by selecting it from the Object List in the category Accessories and Misc. Hardware > Touchpanels. Alternatively select and drag and drop it to the worksheet from the window Accessories and Misc. Hardware > Touchpanels.
The following dialog appears:

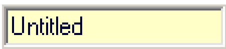


Please enter the number of devices and the Communication Interface and confirm with the OK button. After that one or several TPI-5 Devices appear on the worksheet. The TPI-5 devices can be selected and placed in the worksheet. By right clicking on a TPI-5 Device > Configuration you get to the Configuration Dialog.

TPI-5 Configuration Dialog

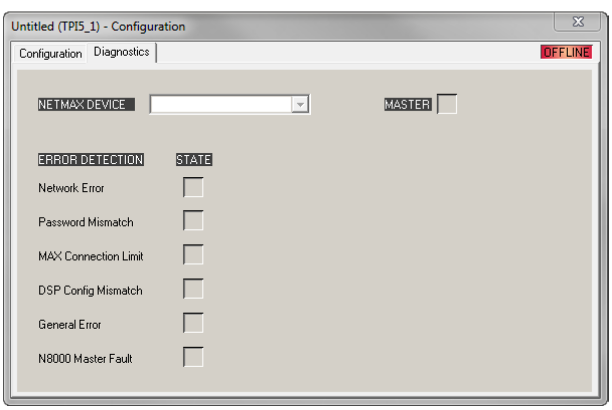
Dialog	Description
Configuration	In this dialog the graphical user interface of the TPI-5 can be designed. Several standard IRIS-Net Controls like e.g. Switch Button, Fader or LED are available.
Diagnostics	On this page different fault states are shown.

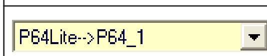


Element	Description
	By clicking this button, the graphical user interface will be exported as a project file (*ds)
	Enter here the IP address matching the hardware IP address. The default IP address of the TPI-5 is 192.168.1.102.
	A name can be assigned to each touch panel to specify its use or position. Click on the yellow entry field and enter the desired name. Press Return on the keyboard to acknowledge the entered name.

Following properties can be used for advanced configuration of the TPI-5:

Property	Description
Brightness	Allows adjusting the display brightness (0 = dark, 100 = bright)
Layer after screensaver active	Select the layer to be shown after activation of the screensaver.
Startup layer	Select the layer to be shown after power-on of the TPI-5.



Element	Description
	Select one of the Electro-Voice NetMax N8000 or DYNACORD P 64 operated by the TPI-5.
MASTER	Master-Fault-Flag for all N8000 / P 64 used on the TPI-5 and the TPI-5 itself.
Network Error	Network Error between a TPI-5 and a connected N8000 / P 64.
Password Mismatch	If the password is wrong the TPI-5 can not connect to a N8000 / P 64.
MAX Connection Limit	If too many users are connected to the N8000 / P64, the TPI-5 can not connect to the N8000 / P64.
DSP Config Mismatch	If the DSP structure saved in the TPI-5 is not identical with the DSP Structure of the N8000 / P64, then the TPI-5 can't go online with the N8000 / P 64.
General Error	General Error in the TPI-5.
N8000 Master Fault	The Master Fault Flag of a N8000 / P64, which is connected to the TPI-5, is active.

Editing TPI properties

Following table lists properties of the TPI-5 touch panel.

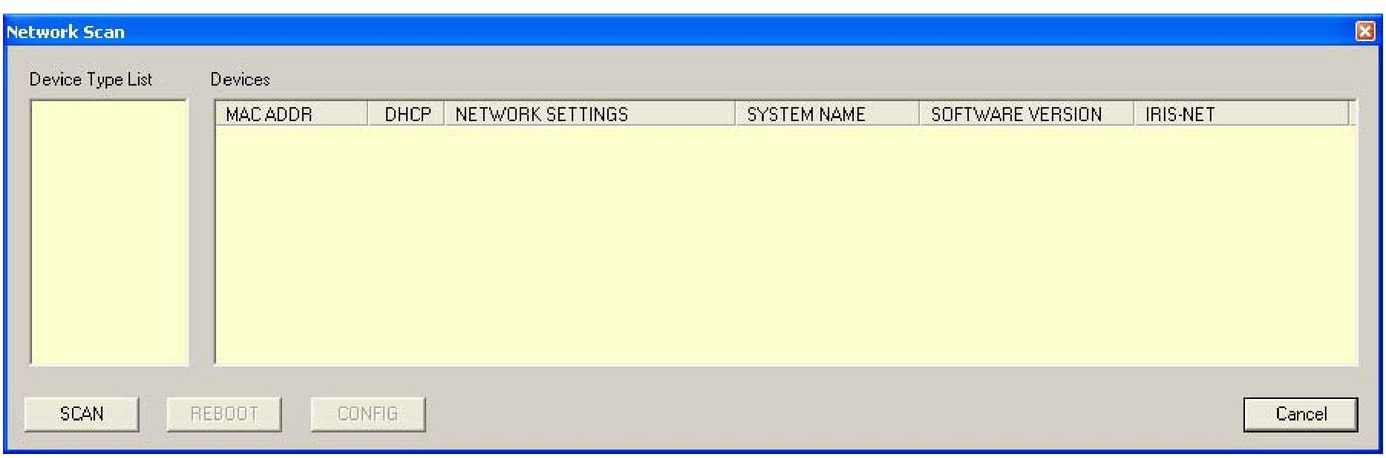
Property	Range	Description
Brightness	0 to 100	Display brightness, 0 = minimum brightness, 100 = maximum brightness
Layer after screensaver active	1 to 32	Enter the number of the layer to be displayed after the screen saver was active.
Startup layer	1 to 32	Enter the number of the layer to be displayed after power-on.

Editing network settings

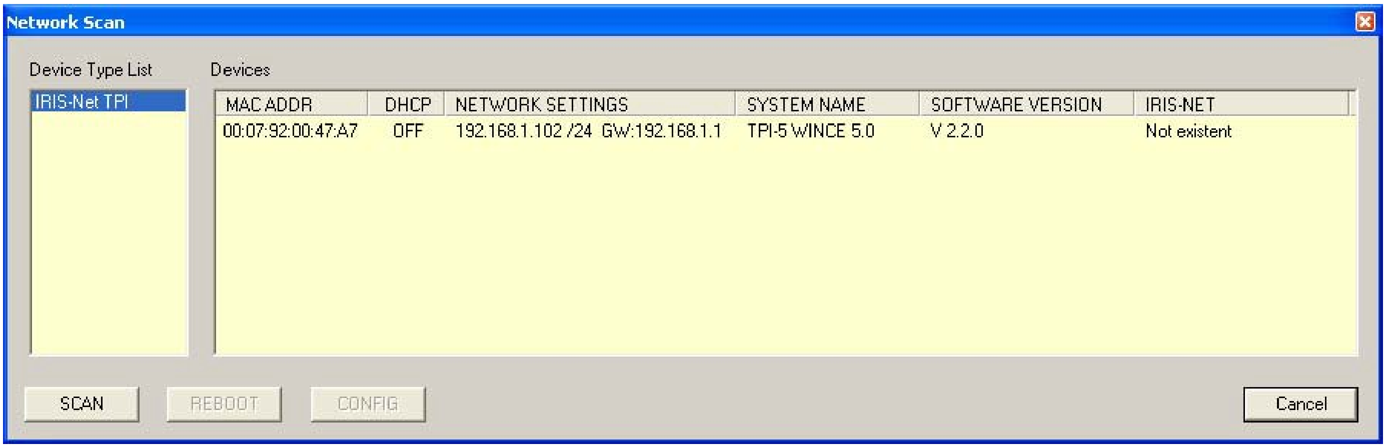
Default network settings of the TPI-5:

Parameter	Value
IP address	192.168.1.102
Subnet mask	255.255.255.0
Gateway IP address	192.168.1.1
DHCP server	off

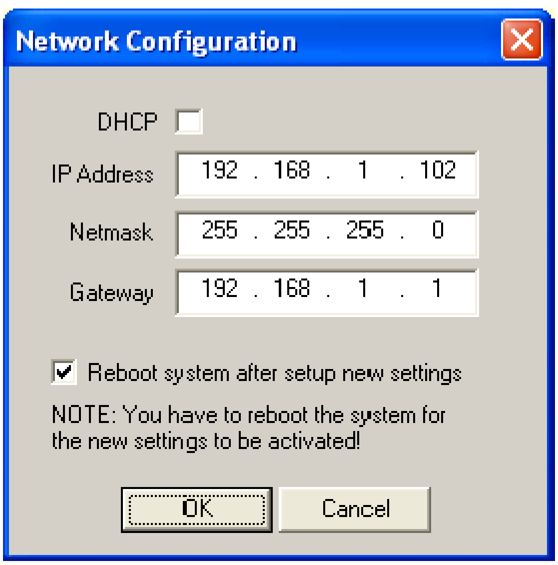
1. Connect the network connection of your PC to the Ethernet interface of the TPI-5 with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-5.
   After some seconds an IRIS-Net project signals the successful start activity of the TPI-5.
3. Start the IRIS-Net software on your PC.
4. Click on Tools > Device Scan.
   The Device Scan dialog appears.



5. Select the entry IRIS-Net TPI in the Device Type list.
   In der Devices Liste werden alle gefundenen Touch Panels angezeigt.



6. By double-clicking on the TPI-5 Device which's network settings you want to change the Network Configuration dialog opens.



7. Change the network settings and confirm with the OK button.

The TPI-5 takes the new settings and reboots.

Updating the IRIS-Net project file

Usually the project file of the TPI-5 is transferred to the TPI-5 during the Going Online procedure shown on the Go Online Dialog. Alternatively a project file can be exported from the Configuration Dialog of a TPI-5 and then be updated as described below. In the following it is assumed that the file to be transferred is available at the PC and the network settings are set to factory defaults.

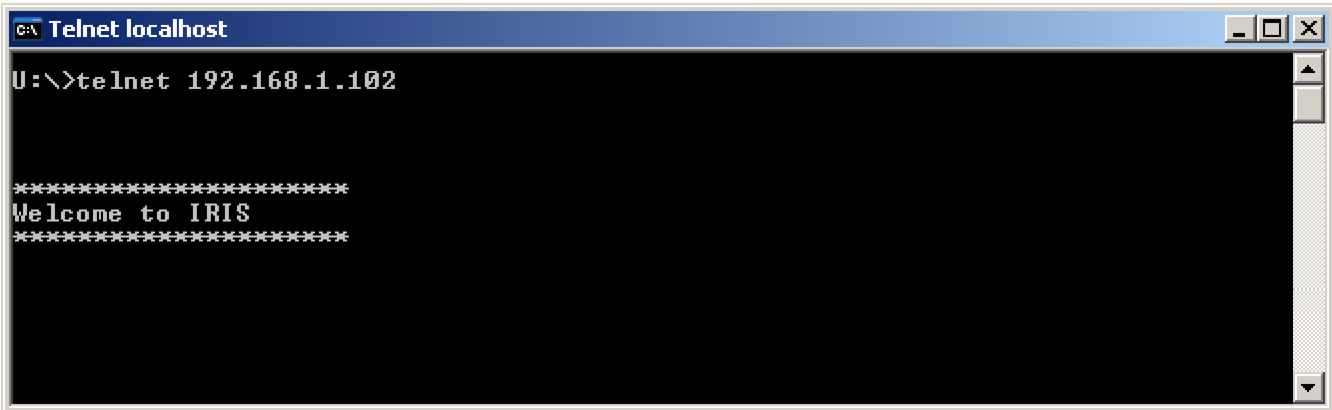
Caution!

Project files including Dante configuration can not be used with TPI-5 touch panel.

Consequences

1. Connect the network connection of your PC to the Ethernet interface of the TPI-5 with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-5.
   After some seconds a IRIS-Net project signals the successful start activity of the TPI-5.

3. On your PC click on Start > All Programs > Accessories > Command Prompt.
   The window Command Prompt appears.
4. . Enter telnet 192.168.1.102 and tap the return button.
   The message "Welcome to IRIS" is indicated.



5. Enter doc*update=start and tap the return button.

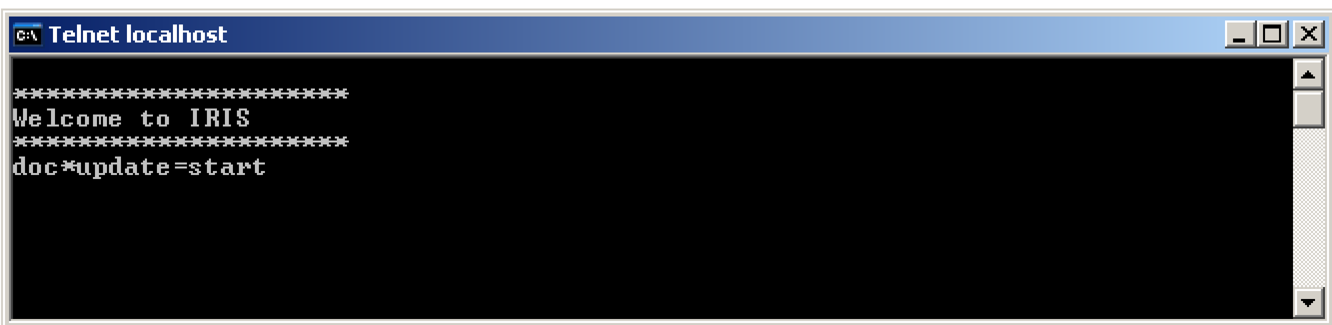
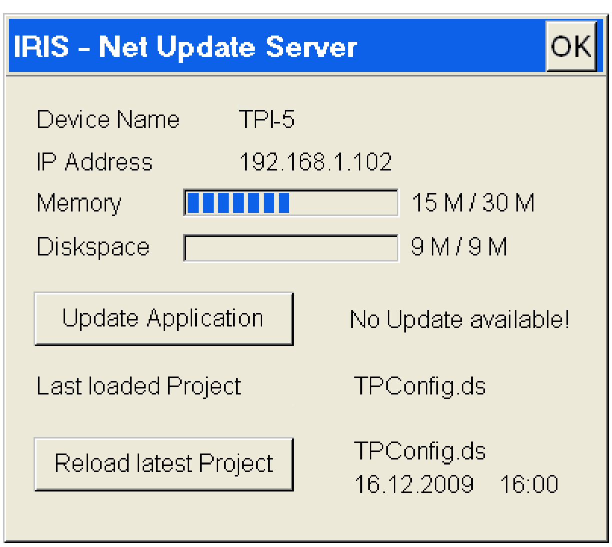
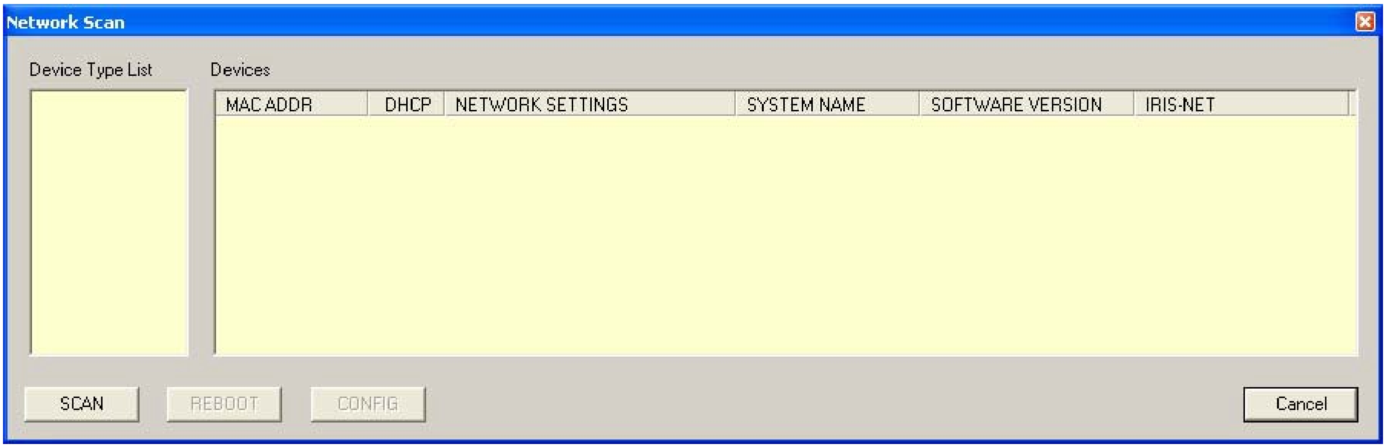


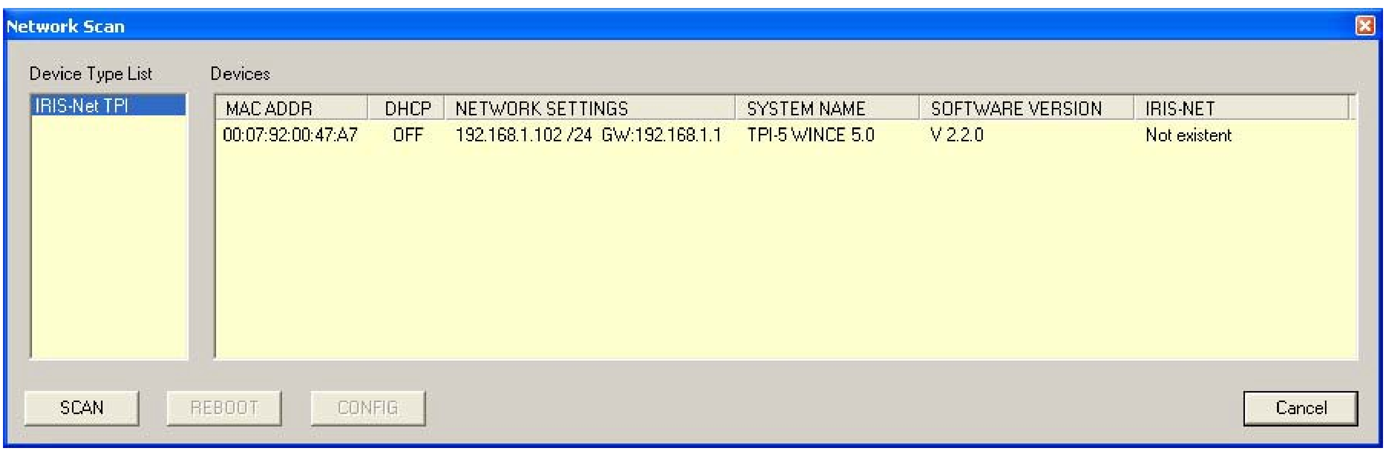
Figure 5.1: The IRIS-Net Update Server dialog appears on the screen of the TPI-5. The TPI-5 is now ready to receive.



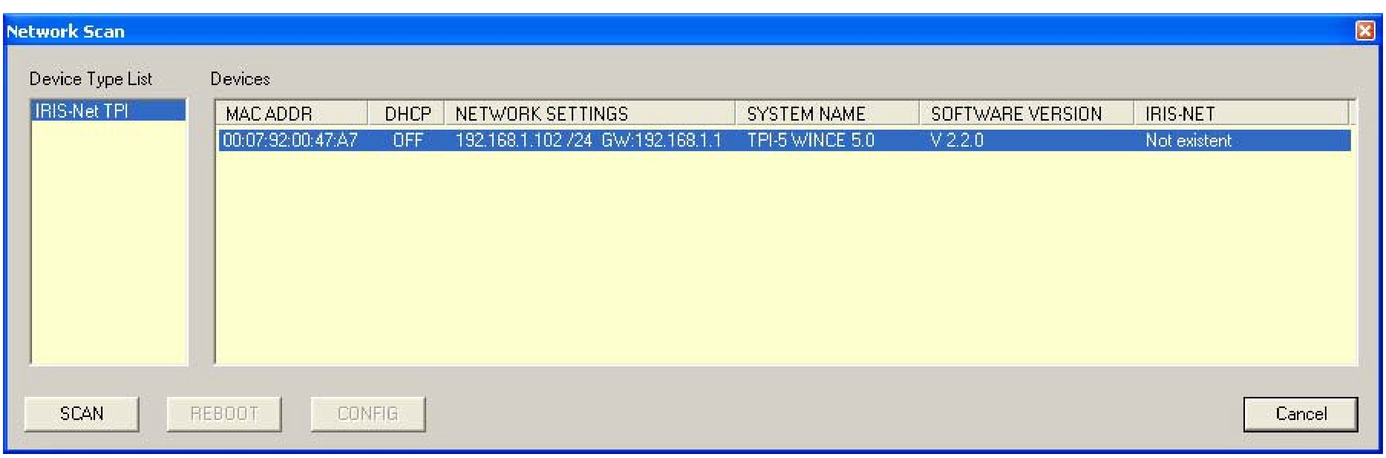
6. Start the IRIS-Net application on your PC.
7. Click on Tools > Device Scan.
   The dialog Device Scan appears.



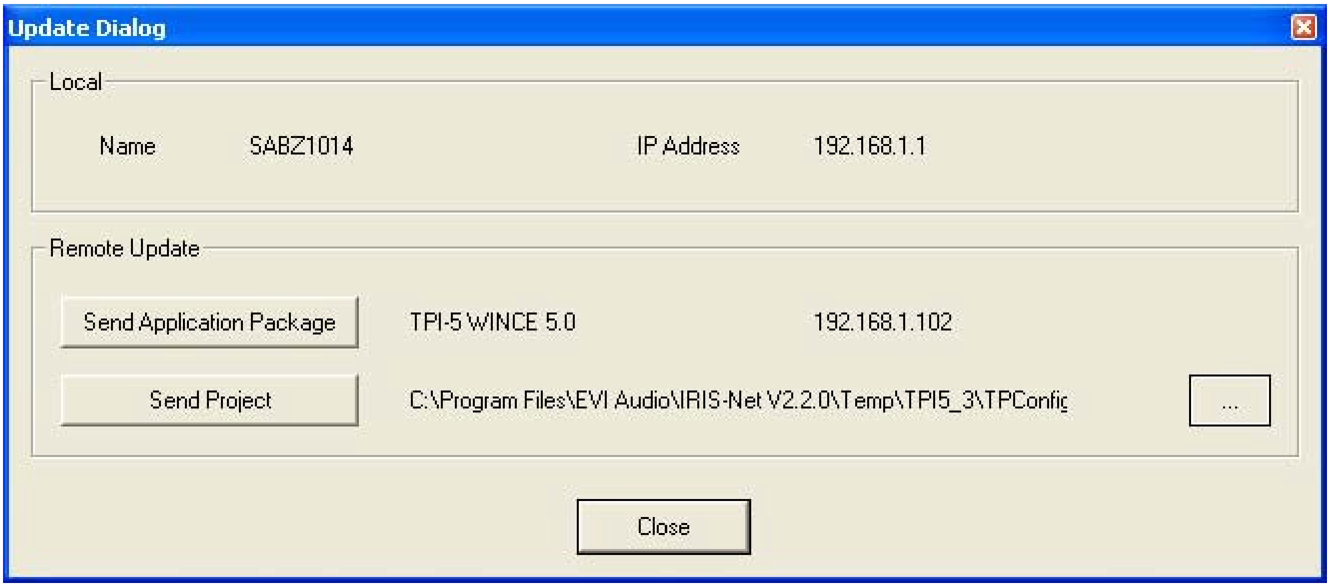
8. Select the entry IRIS-Net TPI in the Device Type list.
   All connected Touch Panels are shown in the Devices list.



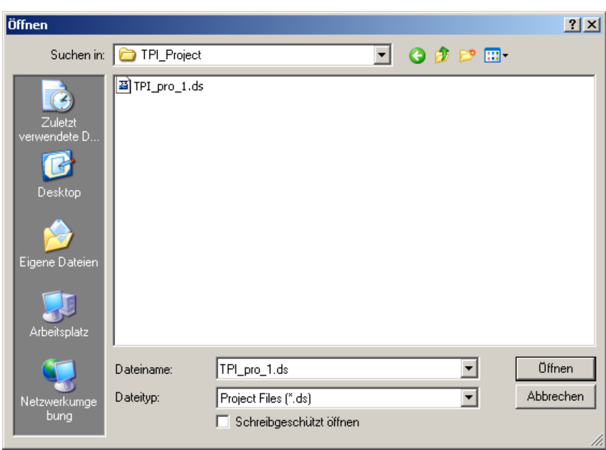
9. Select the Touch Panel from the Devices list that has to be updated.



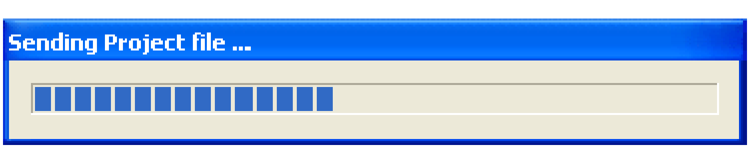
10. Right click on the entry in the Devices list.
    The Update Dialog appears.



11. In the Update Dialog click on button...
    The window Open appears.



12. Select the file to be transferred in the window Open and click on the Open button.
    The file type Project Files (*.ds) can be selected.
    HINT: The project file must be renamed to "TPConfig.ds" before sending it, so that after a reboot of the TPI-5 it can automatically be opened.
13. Click on button Send Project in the Update Dialog.
    The file is now sent to the TPI-5. During the transmission a progress bar will pop up.
    After successful transmission the name of the transmitted project file is indicated in the IRIS-Net Update Server dialog on the touch panel.



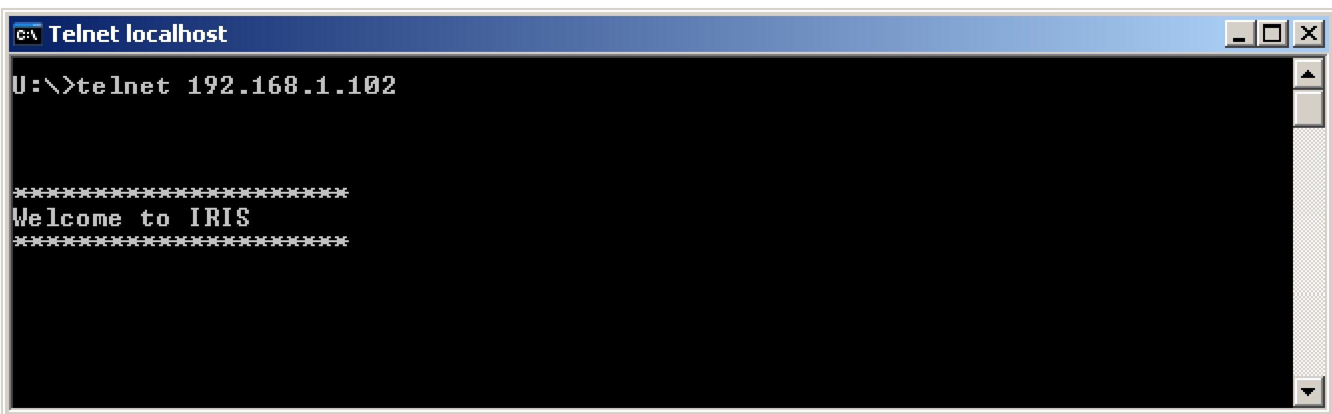
14. Click on button Reload latest Project in the IRIS-Net Update Server dialog at the Touch Panel. .
    The new project is loaded

15. Check the new project on the Touch Panel.
16. Click on button OK in the IRIS-Net Update Server dialog on the Touch Panel.

Updating the IRIS-Net application file

The purpose of this procedure is to build a connection between a PC and a TPI-5 and updating the IRIS-Net project file of the TPI-5. In the following it is assumed that the file to be transferred is available at the PC and the network settings are set to factory defaults.

1. Connect the network connection of your PC to the Ethernet interface of the TPI-5 with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-5.
   After some seconds a IRIS-Net project signals the successful start activity of the TPI-5.
3. On your PC click on Start > All Programs > Accessories > Command Prompt.
   The window Command Prompt appears.
4. Enter telnet 192.168.1.102 and tap the return button.
   The message "Welcome to IRIS" is indicated.



5. Enter doc*update=start and tap the return button.

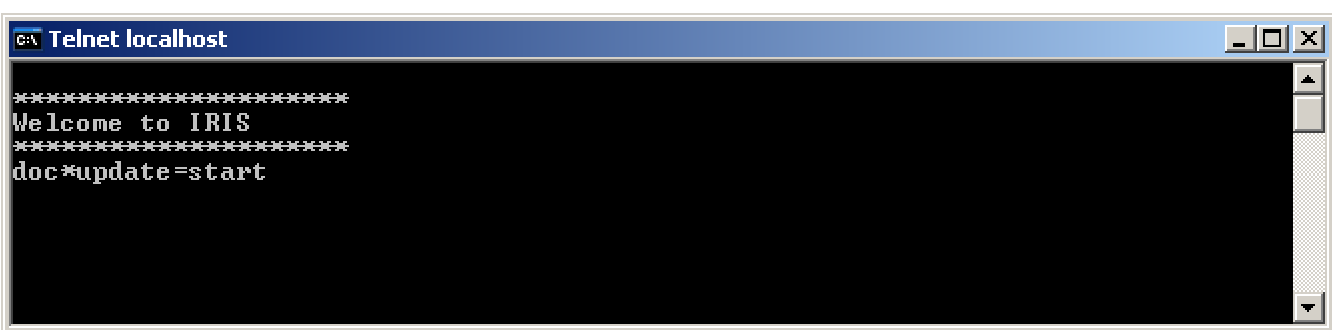
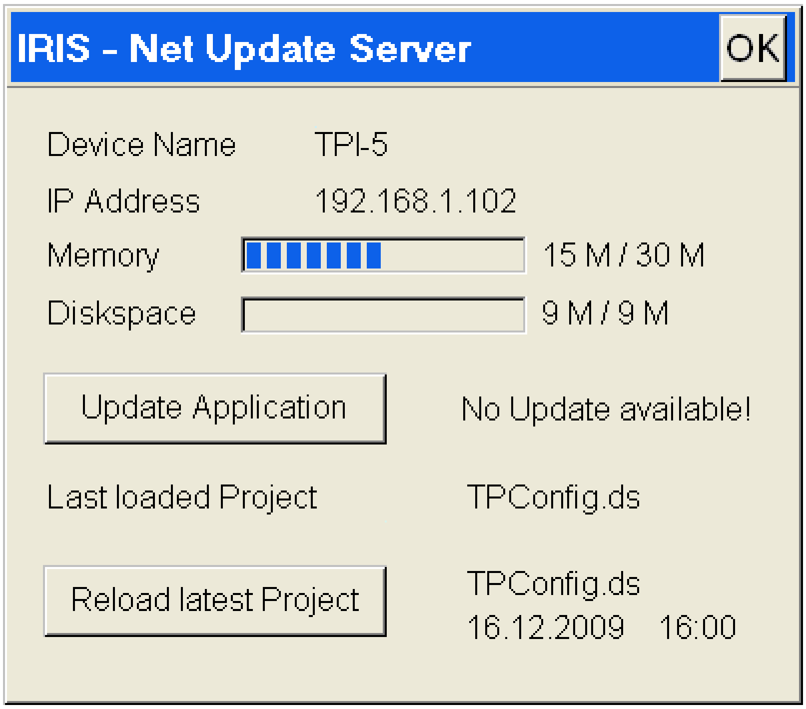
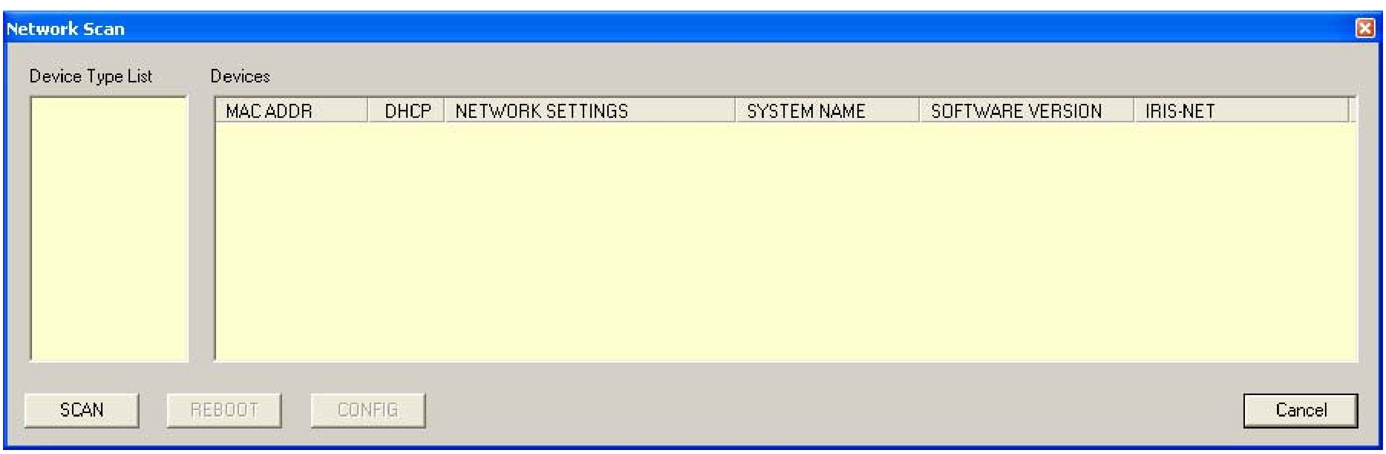


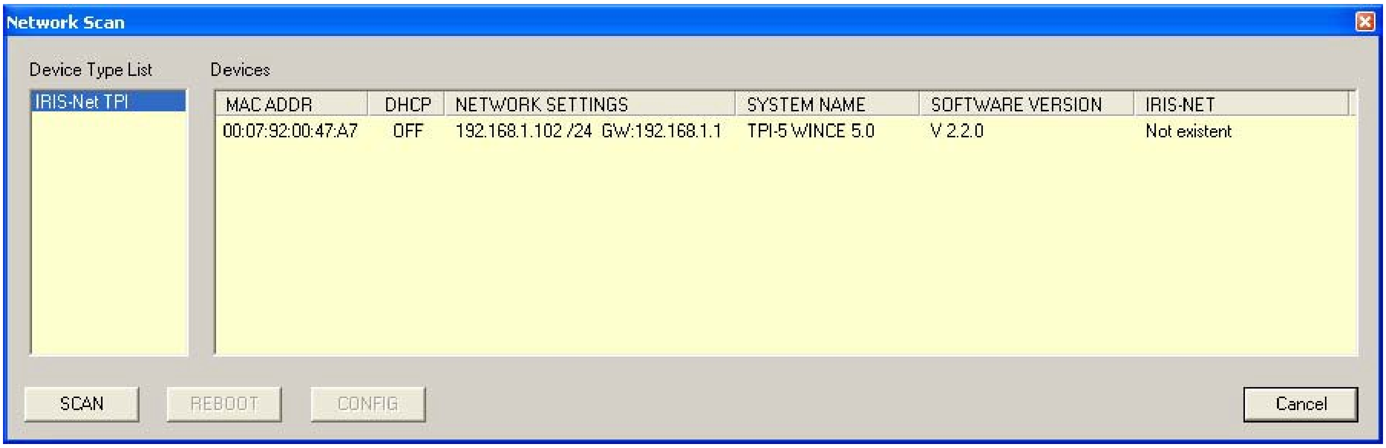
Figure 5.2: The IRIS-Net Update Server dialog appears on the screen of the TPI-5. The TPI-5 is now ready to receive.



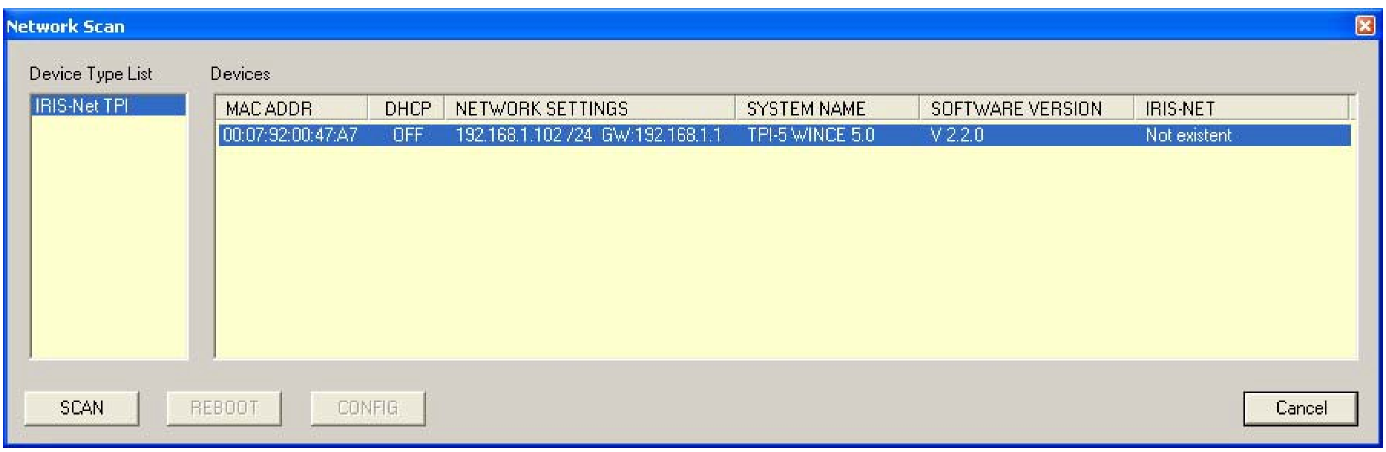
6. Start the IRIS-Net application on your PC.
7. Click on Tools > Device Scan.
   The dialog Device Scan appears.



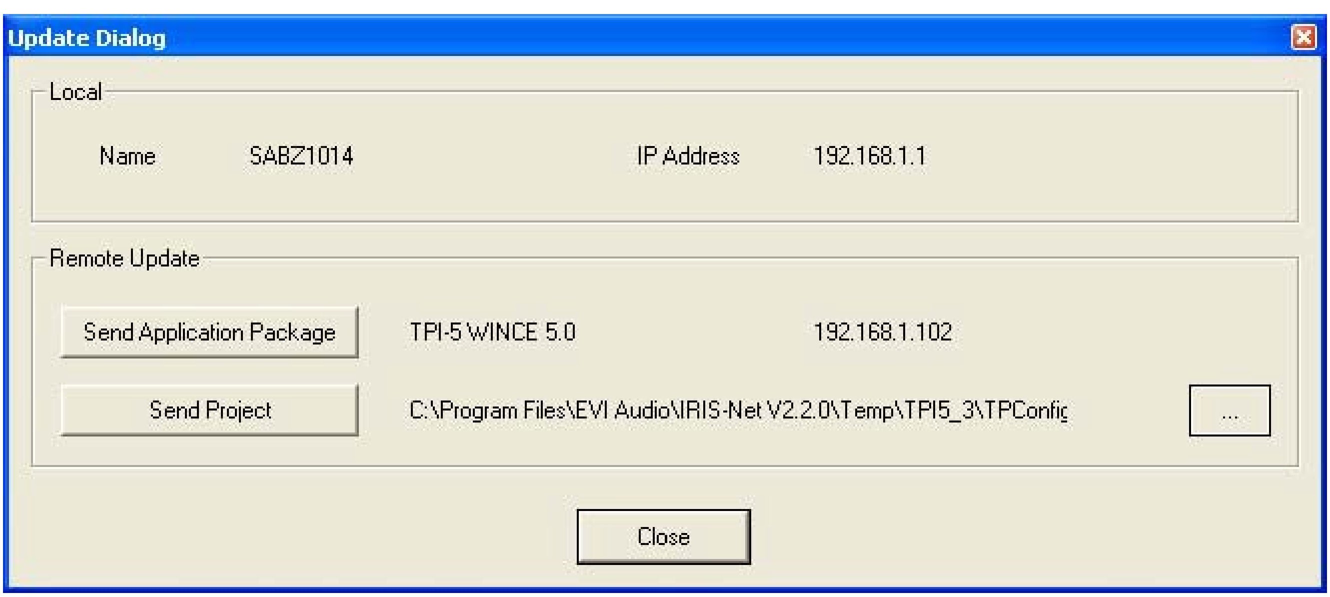
8. Select the entry IRIS-Net TPI in the Device Type list.
   All connected Touch Panels are shown in the Devices list.



9. Select the Touch Panel from the Devices list that has to be updated.



10. Right click on the entry in the Devices list.
    The Update Dialog appears.



11. Click on the button Send Application Package in the Update Dialog.
    The application file is now sent to the TPI-5. During the transmission a progress bar will pop up.

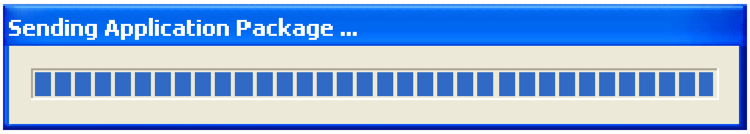
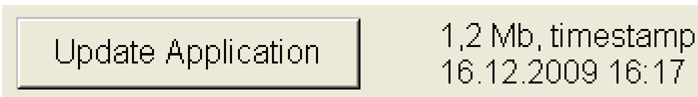


Figure 5.3: After successfull transmission the file size and date of the application file is indicated in the IRIS-Net Update Ser- ver dialog on the TPI-5.

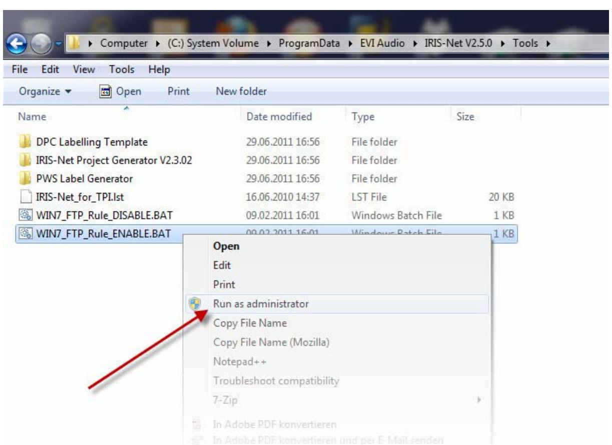


12. Click on button Update Application in the IRIS-Net Update Server dialog on the Touch Panel. The new Application Package will be extracted and installed. The new project will be loaded automatically.
13. Click on button OK in the IRIS-Net Update Server dialog on the Touch Panel.

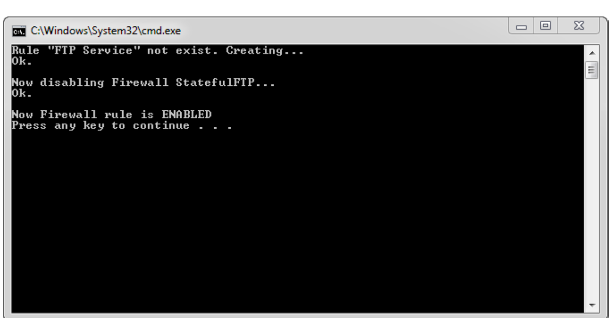
Windows 7

When using Windows 7 the integrated firewall has to be configured for using the TPI-5. The following section describes how to configure the firewalll before and after updating the project file or IRIS-Net application.

1. Open the directory /Tools in the IRIS-Net installation directory.
2. Open the context menu of the file WIN7_FTP_Rule_ENABLE.BAT using the right mouse button and click on the “Run as administrator” entry.

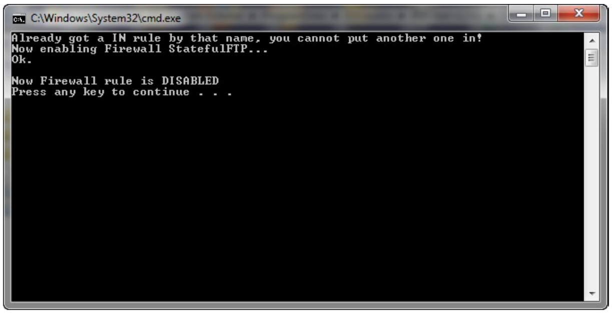


3. If requested, enter the administrator password of Windows 7. A DOS window will appear indicating the successful configuration of the firewall.



4. Now, update the project file or IRIS-Net application as described in the corresponding chapters.
5. Open the context menu of the file WIN7_FTP_Rule_DISABLE.BAT using the right mouse button and click on the “Run as administrator” entry.

6. If requested, enter the administrator password of Windows 7. A DOS window will appear indicating the successful re-configuration of the firewall.



TPI-8/TPI-12

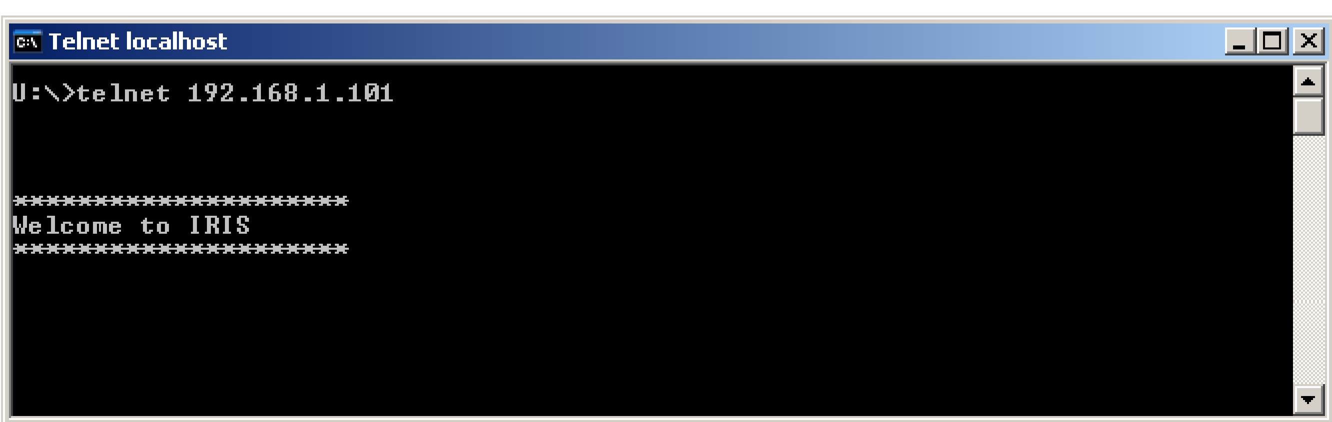
Updating the IRIS-Net project file

HINT: The following description refers to IRIS-Net version V2.0 or newer. The IRIS-Net version installed at the TPI must be V1.8.0 or newer. For older version of IRIS-Net (e.g. V1.7.1) please see page 404.

Caution!
Project files including Dante configuration can not be used with TPI-8/TPI-12 touch panels.
Consequences

The purpose of this procedure is to build a connection between a PC and a TPI-8/TPI-12 and updating the IRIS-Net project file of the TPI-8/TPI-12. In the following it is assumed that the file to be transfered is available at the PC.

1. Connect the network connection of your PC to the Ethernet interface of the TPI-8/TPI-12 directly with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-8/TPI-12.
   After some seconds an IRIS-Net project signals the successful start activity of the TPI-8/TPI-12.
3. Click on Start > All Programs > Accessories > Command Prompt.
   The window command prompt appears.
4. Enter telnet 192.168.1.101 and tap the return button.
   The message “Welcome to IRIS” is indicated.



5. Enter doc*update=start and tap the return button.

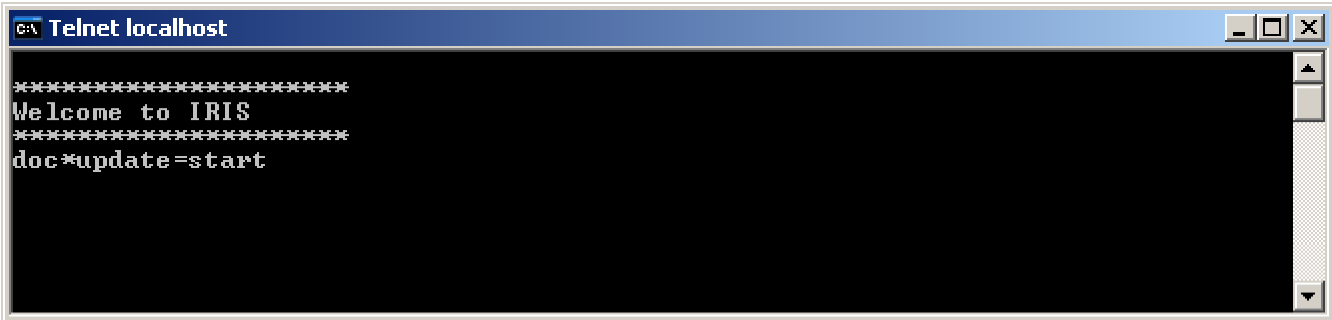
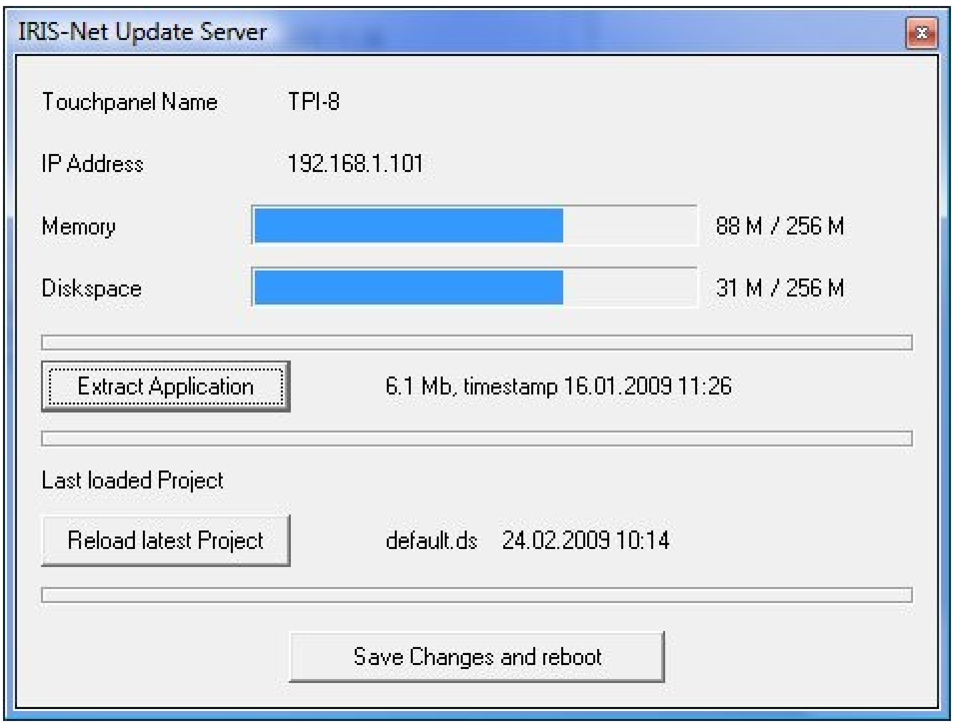
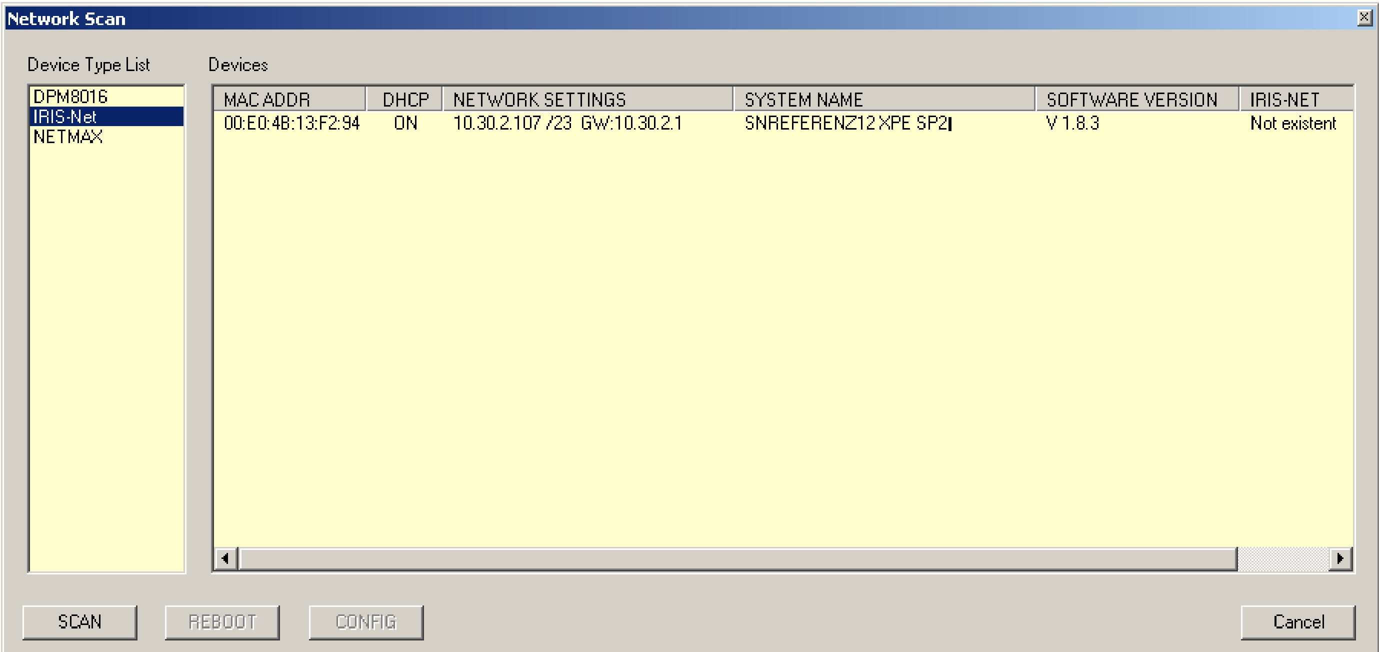


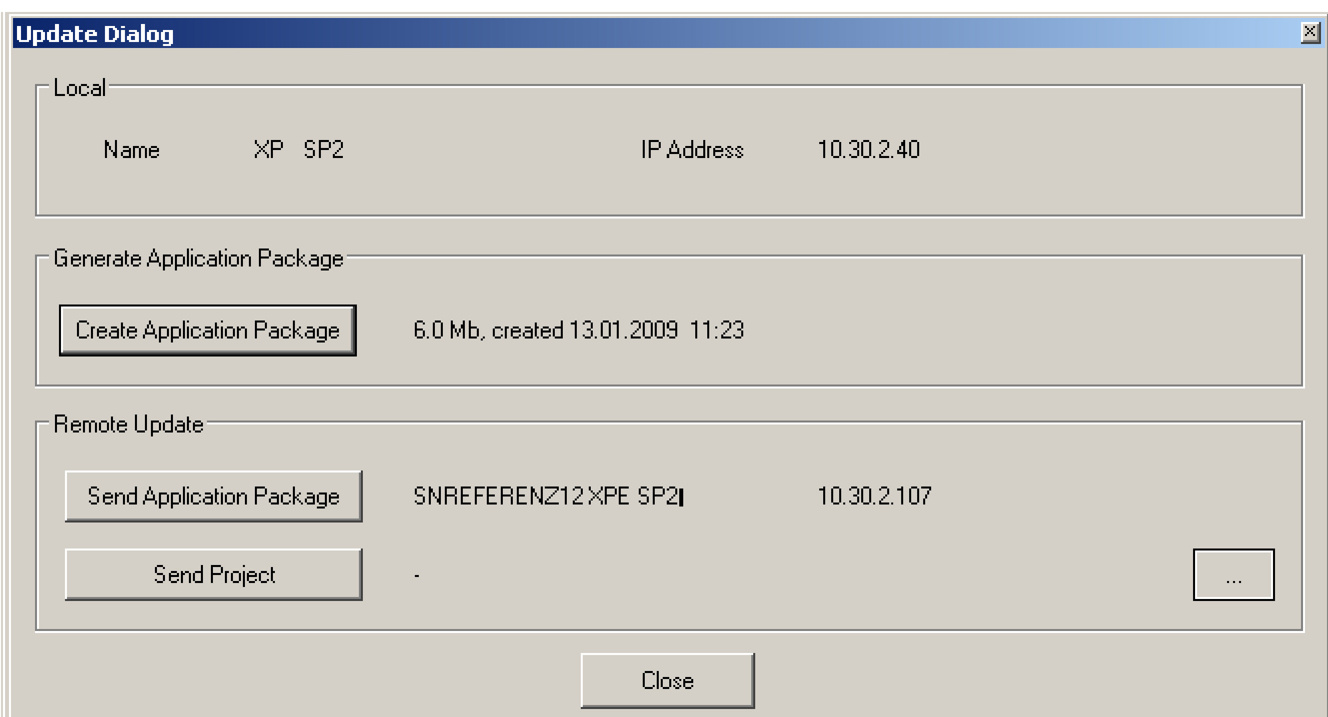
Figure 5.4: The dialog IRIS-Net Update Server appears at the screen of the TPI-8/TPI-12. The TPI-8/TPI-12 is now ready-to- receive.



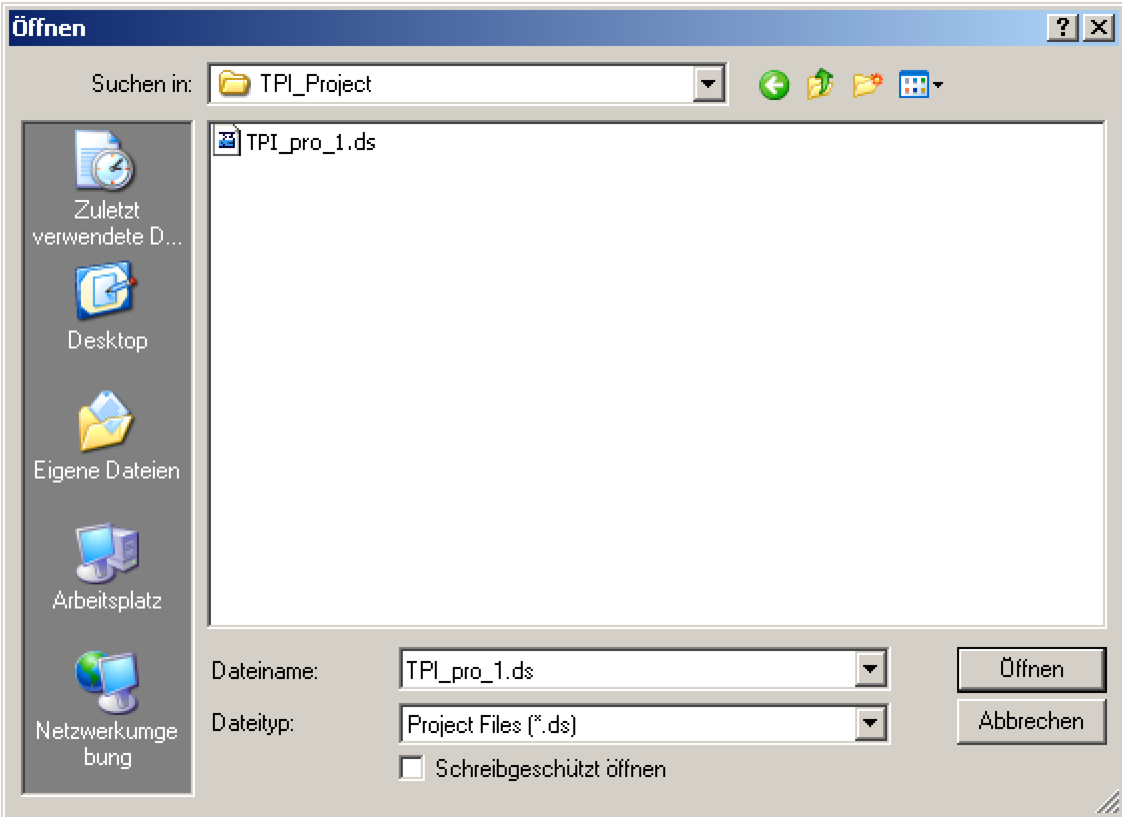
6. Start IRIS-Net application on your PC.
7. Click on Tools > Device Scan.
   The dialog Device Scan appears.



8. Select the entry IRIS-Net in the Device Type list.
   All connected Touch Panels are shown in the Devices list.
9. Select the Touch Panel from the Devices list that has to be updated.
10. Right click on the entry in the Devices list.
    The dialog Update Dialog appears.



11. Click on button...in the Update Dialog dialog.
    The window Open appears.



12. Select the file to be transferred in the window Open and click on the Open button.
    The file type Project Files (*.ds) can be selected.
13. Click on button Send Project in the Update Dialog dialog.
    The file is now sent to the TPI-8/TPI-12. The successful transmission is indicated by the name of the project file in the IRIS-Net Update Server dialog at the TPI-8/TPI-12.
14. Click on button Reload latest Project in the IRIS-Net Update Server dialog at the Touch Panel.
    The new project is loaded.

15. Check the new project at the Touch Panel.
16. If the new project works, click on button Save changes + Reboot in the IRIS-Net Update Server dialog..

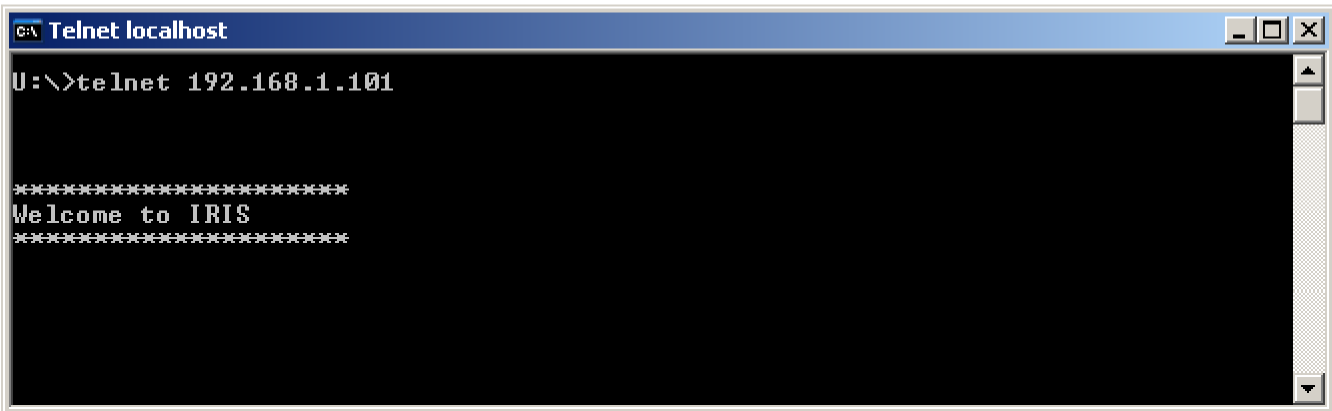
17. To restore the previous project file, turn off the power supply system of the Touch Panel. The old project will be loaded when switching on the Touch Panel next time.

Updating the IRIS-Net application file

HINT: The following description refers to IRIS-Net version V2.0 or newer. The IRIS-Net version installed at the TPI must be V1.8.0 or newer. For older version of IRIS-Net (e.g. V1.7.1) please see page 404.

The purpose of this procedure is to build a connection between a PC and a TPI-8/TPI-12 and updating the IRIS-Net project file of the TPI-8/TPI-12. In the following it is assumed that the file to be transferred is available at the PC.

1. Connect the network connection of your PC to the Ethernet interface of the TPI-8/TPI-12 directly with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-8/TPI-12.
   After some seconds a IRIS-Net project signals the successful start activity of the TPI-8/TPI-12.
3. Click on Start > All Programs > Accessories > Command Prompt.
   The window command prompt appears.
4. Enter telnet 192.168.1.101 and tap the return button.
   The message “Welcome to IRIS“ is indicated.



5. Enter doc*update=start and tap the return button.

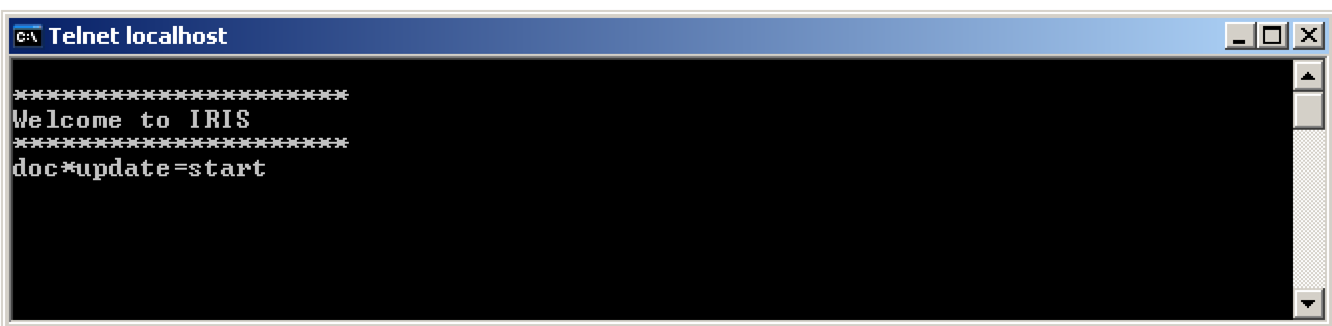
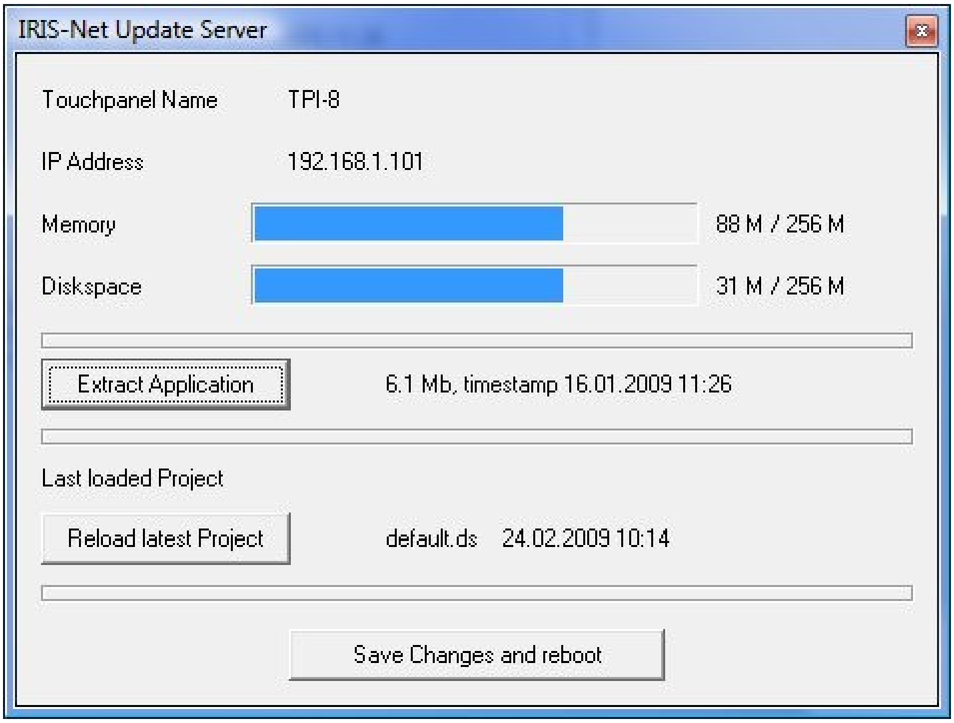
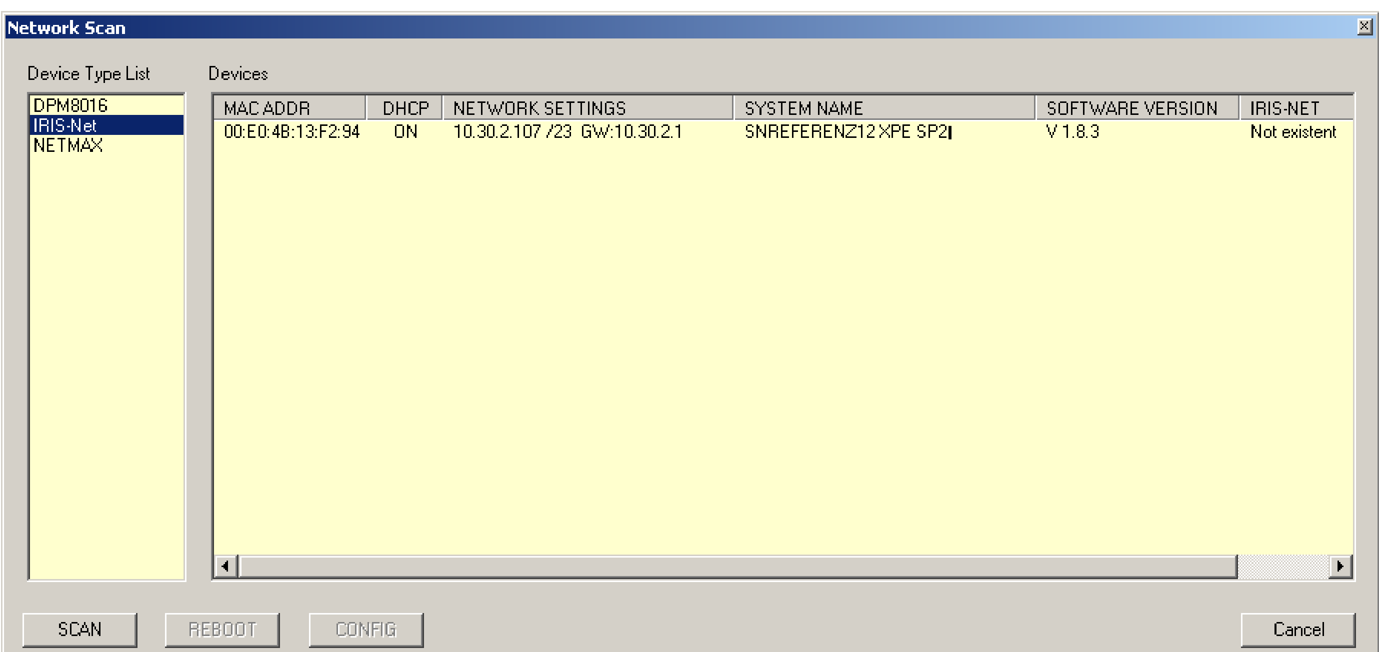


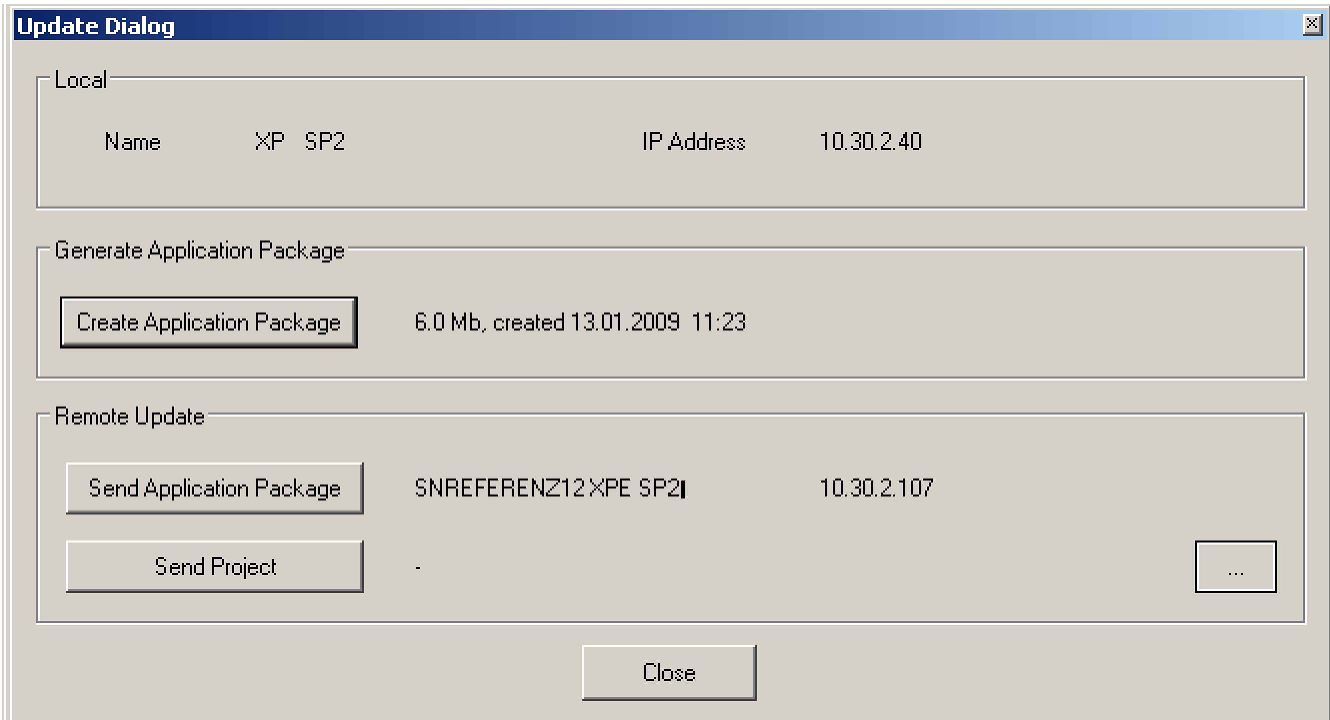
Figure 5.5: The dialog IRIS-Net Update Server appears at the screen of the TPI-8/TPI-12. The TPI-8/TPI-12 is now ready-to- receive.



6. Start IRIS-Net application on your PC.
7. Click on Tools > Device Scan.
   The dialog Device Scan appears.



8. Select the entry IRIS-Net in the Device Type list.
   All connected Touch Panels are shown in the Devices list.
9. Select the Touch Panel from the Devices list that has to be updated.
10. Right click on the entry in the Devices list.
    The dialog Update Dialog appears.



11. If Archive not yet created! is indicated next to the Create Application Package button, click on the Create Application Package button.

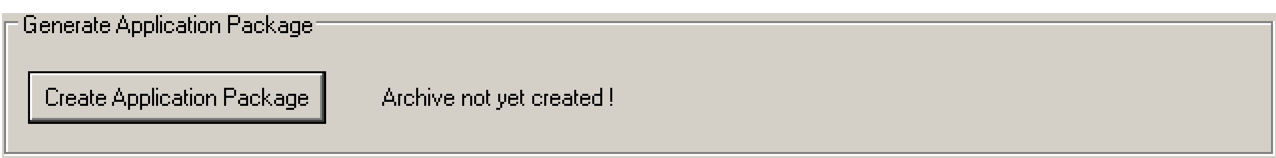


Figure 5.6: An application package is generated from the currently running IRIS-Net application. The file size and date of the generated package is indicated next to the Create Application Package button when finished.

12. Click on the button Send Application Package in the Update Dialog dialog.

Figure 5.7: The application file is now sent to the TPI-8/TPI-12. The successful transmission is indicated by the file size and date of the application file in the IRIS-Net Update Server dialog at the TPI-8/TPI-12.

HINT: If the indicated application file size is not identical at the PC and the Touch Panel, repeat sending the file.

13. Click on button Extract Application in the IRIS-Net Update Server dialog.
    The received Application Package is prepared for installation.

14. Click on button Reload latest Project in the IRIS-Net Update Server dialog.
    The new IRIS-Net version is started and the latest project file is loaded.

15. Check if the project file works in the new IRIS-Net version.
16. If the new IRIS-Net version should be used, click on button Save changes and reboot in the IRIS-Net Update Server dialog.

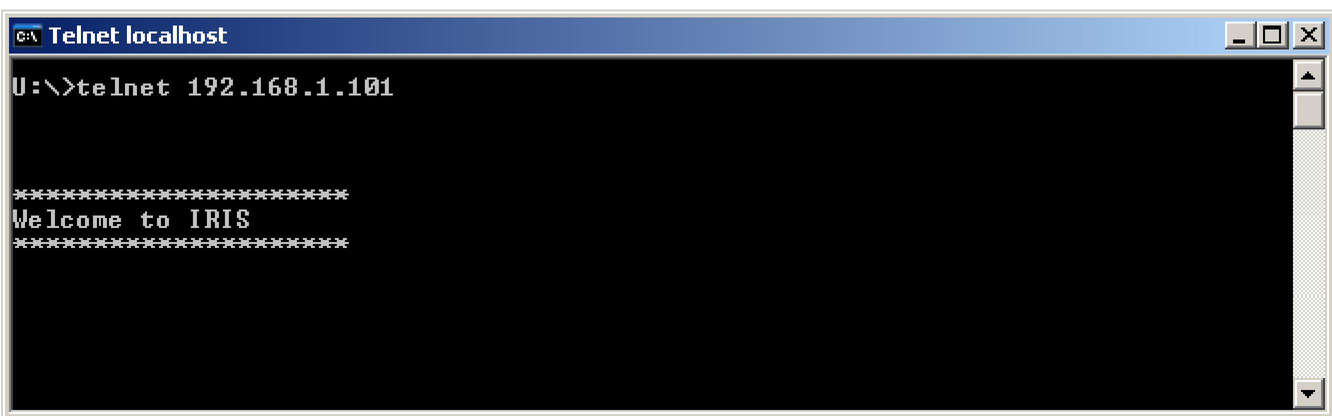
17. To restore the previous version of IRIS-Net, turn off the power supply system of the Touch Panel. The previous version of IRIS-Net will be loaded when switching on the Touch Panel next time.

Hot to update for IRIS-Net V1.8.3 and older

HINT: The following description refers to IRIS-Net versions older than V2.0.

The purpose of this procedure is to build a connection between a PC and a TPI-8/TPI-12 with factory network settings (see *Going On-Line, page 13*) and updating the IRIS-Net project file or IRIS-Net application of the TPI-8/TPI-12. In the following it is assumed that the file to be transferred is available at the PC.

1. Connect the network connection of your PC to the Ethernet interface of the TPI-8/TPI-12 directly with a crossover cable, or with a patch cable and a hub/switch.
2. Activate the power supply system of the TPI-8/TPI-12.
   After some seconds a IRIS-Net project signals the successful start activity of the TPI-8/TPI-12.
3. Click on Start > All Programs > Accessories > Command Prompt.
   The window command prompt appears.
4. Enter telnet 192.168.1.101 and tap the return button.
   The message “Welcome to IRIS” is indicated.



5. Enter doc*update=start and tap the return button.

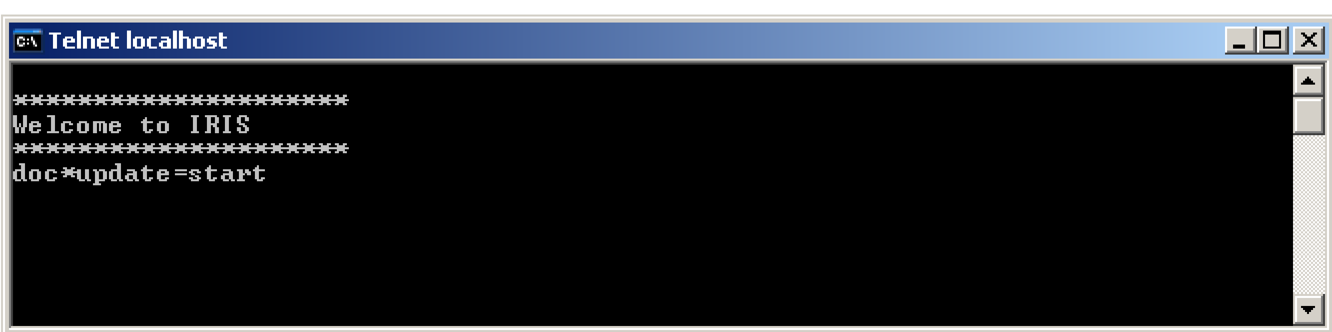
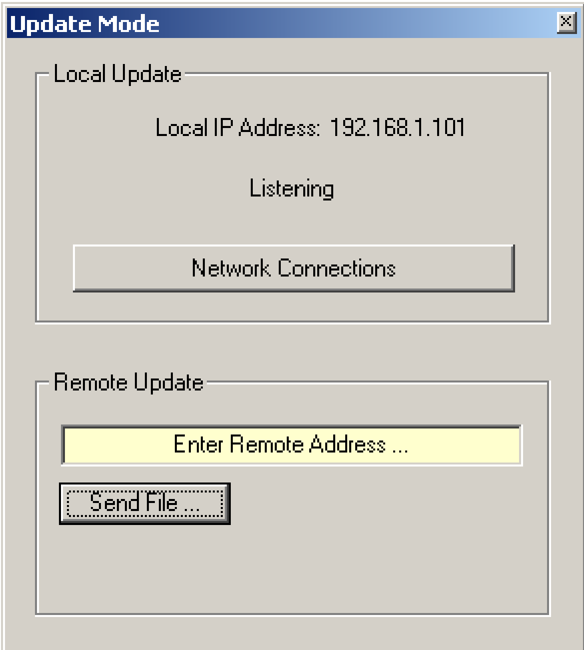
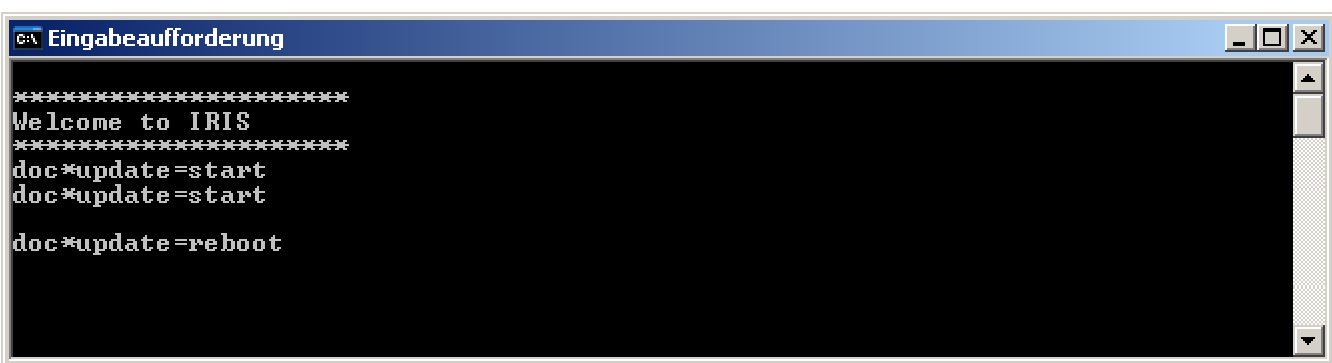


Figure 5.8: The dialog Update Mode and a onscreen keyboard appears at the screen of the TPI-8/TPI-12. The TPI-8/TPI-12 is now ready-to-receive.

6. Start IRIS-Net application on your PC.
7. Click on Configuration > Update Touch Panel.
   The dialog Update Mode appears.



8. Type 192.168.1.101 in the Enter Remote Address.. input field and tap the return button.
9. Click on Send File...button in the Update Mode dialog.
   The window Open appears.
10. Select the file to be transferred in the window Open and click on the Open button. The file types Project Files (*.ds) and Application Archives (*.zip) can be selected.
    The file is now sent to the TPI-8/TPI-12. The successful transmission is indicated by message “success” in the Update Mode dialog at the PC and also the TPI-8/TPI-12.
11. Enter doc*update=reboot and tap the return button.
    The TPI-8/TPI-12 reboots and uses the new IRIS-Net project file or IRIS-Net application.



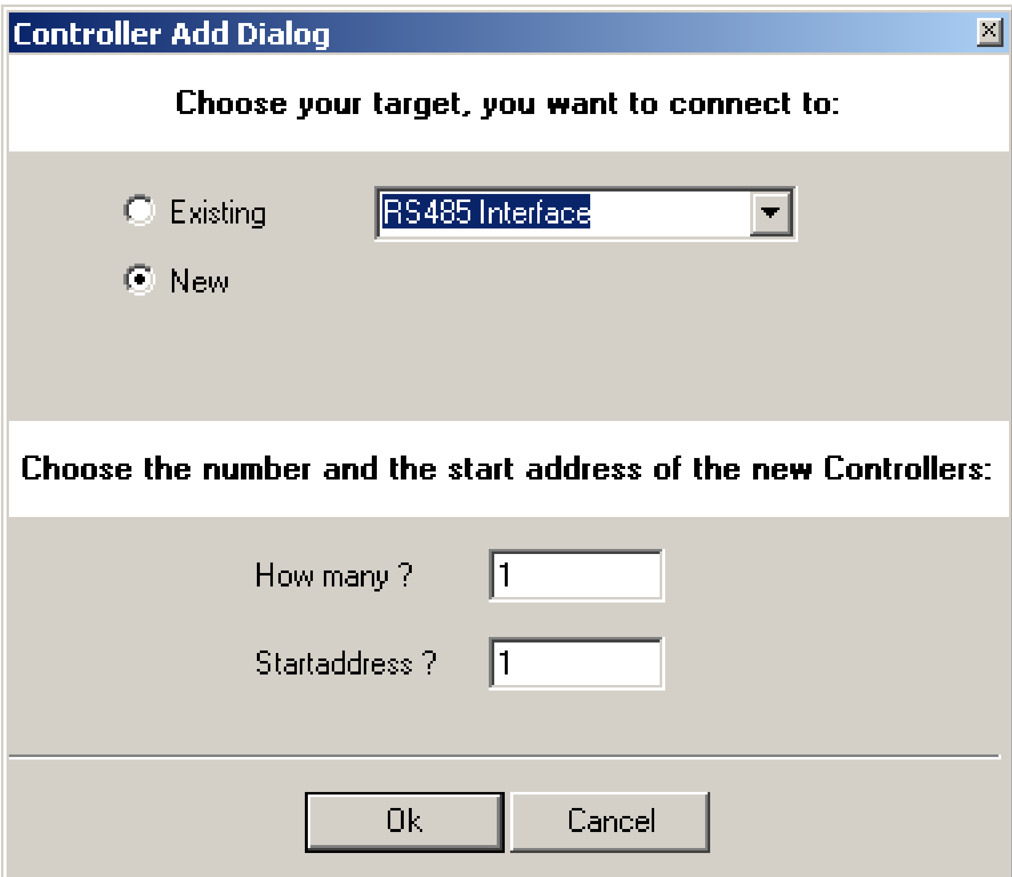
DIGITAL SOUND PROCESSOR

DX38 Digital Sound Processor

The Dx38 is an universal Digital Sound System Processor that provides 2 inputs and 4 outputs; plus internal summing of the inputs 1 and 2. Via Matrix it is possible to assign the outputs to any input or to the sum of the inputs. It is further possible to establish the following configurations: Stereo or Dual 2-Way systems, 3-Way + Direct and 4-Way systems, each with Mono Sub-channel, but also full range systems.
High and low-pass filters are provided for the frequency crossover functions in all operation modes. The selection includes Linkwitz-Riley, Butterworth and Bessel type filters with switchable slopes between 6, 12, 18 and 24 dB/oct. A huge number of additional filters offers extremely flexible correction of the frequency response. Each input incorporates a 5- band equalizer, allowing to assign high and low-pass, high and low-shelving or parametric peak-dip filters to its individual filter sections. Next to the frequency crossover filters, four additional filters are employed in each output channel, which also can be set to work as high or low-pass, high or low-shelving filters, parametric peak-dip filters, or all-pass filters. Additional filtering is provided through 2. order high-passes for the realization of B-6 alignment, or special LPN- filters (Low-Pass Notch filters) for correcting the frequency and phase responses of optimally vented woofer cabinets. Each channel additionally provides a delay, a polarity switch, a programmable level control and a digital compressor / limiter while the master delays are located in the input channels.
The user can choose between two operation modes: the ”No Edit Mode” allows to simply select the required combination of loudspeaker systems from the factory preset program list. Afterwards, the appliance is optimally matched to the sound system and can be operated instantly. The ”Full Edit Mode” on the other hand offers access to all parameters, allowing to freely program and store basically any setting. A total number of 80 memory addresses - 50 preset and 30 freely assignable user-programs - are available.
Within the Dx38, AD/DA conversion is taken care of by linear 24-bit converters; where the AD-section employs 128 times oversampling, gain-ranging Sigma-Delta converters. The DA-section offers 128 times oversampling Sigma-Delta converters. The overall signal processing is performed by two 24-bit Motorola signal processors.
Additional Features are:
– FLASH memory for software and preset updates via serial interfaces
– PC-based operation and configuration software IRIS-Net
– Standard MIDI-interface and RS-232 interface
– RS-485 interface or switching contacts optionally available
– Back-lit graphic-display with 122 x 32 dots
– Inputs and outputs are electronically balanced, XLR-type connectors
– Input transformer-balancing is optionally available
– Input / Output level controls, Output-Mute switch, channel function indicators SUB, LO, MID, HI
– Input / Output meter instruments, compressor and clipping LEDs

Dx38 Device

Start by creating an Dx38 Device in your IRIS-Net project. Drag an Dx38 from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

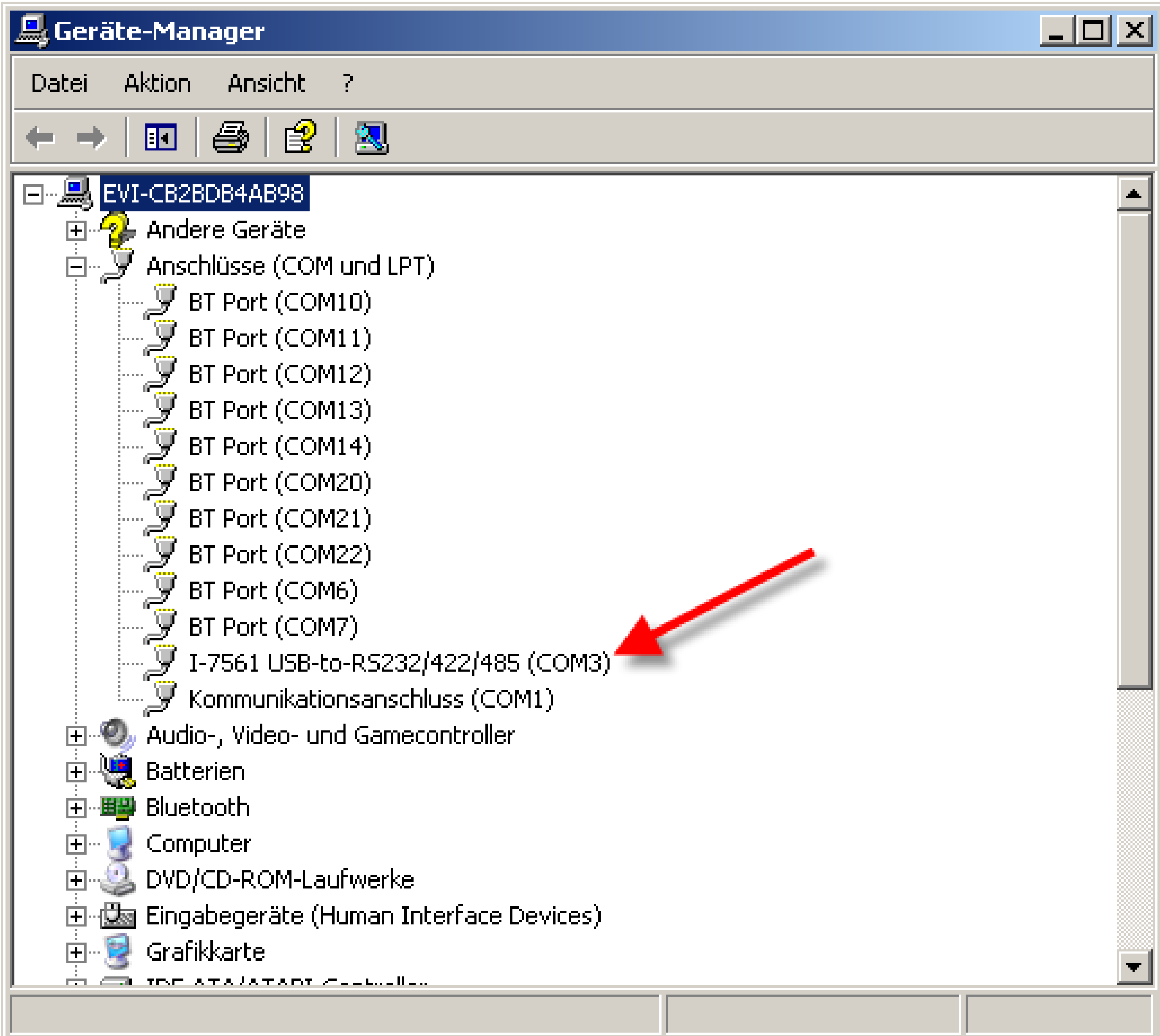


Enter the requried number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of Dx38 Devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the Ctrl key and click on the device. In either case a successfully selected device is shown with a red border around it.

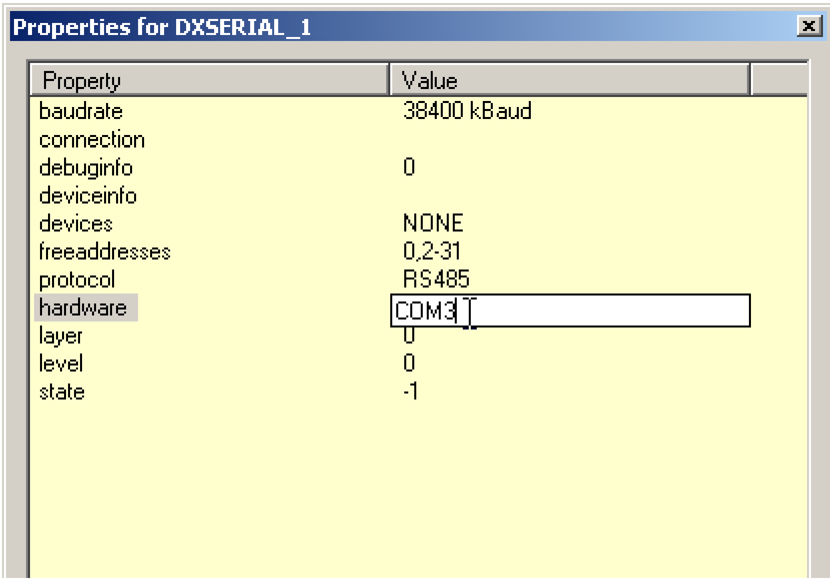
Double clicking on an Dx38 device icon opens the Userpanel.

CONFIGURATION OF THE INTERFACE

For using the Dx38 device in IRIS-Net the used COM port of the PC has to be selected. When using an USB-RS-485 adapters the system internal COM port can be found in the Control Panel, see following picture.



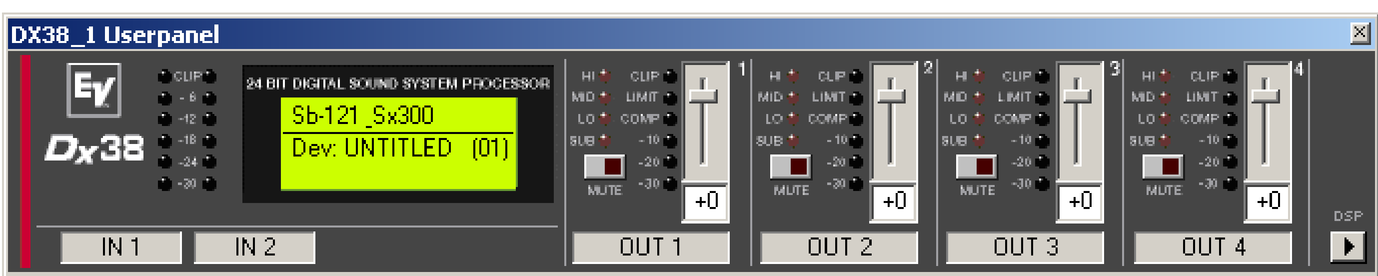
In this example COM port 3 is used by the adapter. The value „COM3” must be entered for the Property „hardware” of the serial interface in IRIS-Net.



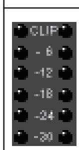
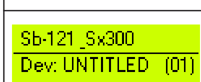
Reference

DX38 USER PANEL

The Dx38 User panel provides access to the controls and indications at the Dx38 front panel. The complete functionality available at the Dx38 LC-Display can be accessed via pressing the DSP button.



Indications and Functions of the Dx38 User panel

Element	Description
	The level meter instruments are meant for optical monitoring of the input signal levels, individually showing the peak value of the correspondent input signal. The input control should be set to a position so that the meter instruments indicate a level between -6 and -12 dB. To prevent inter- nal clipping, make sure that the CLIP LEDs are not lit.
	Line 1 displays the description of the selected User Memory. Line 2 displays the description of the device and its address at the RS-485 bus.
	These LEDs indicate, which frequency band the corresponding channel is set to. If a channel is configured for full range operation, all its func- tion-LEDs are simultaneously lit.

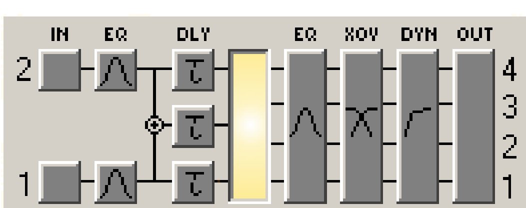
	These keys allow to mute the output signal of the corresponding output channels. Pressing a key once switches the mute-function ON; the key's red LED lights. Pressing the button again switches the mute-function OFF; the key's LED is dimmed again.
	These LEDs indicate the peak level of the corresponding outputs. The Dx38 should be operated in a range, so that the clip-LEDs are not lit. Otherwise, this could lead to internal clipping. The COMP/ LIMIT-LEDs light, when the compressor / limiter of the corresponding channel is activated; e. g.: when the audio signal level exceeded the previously set threshold and therefore the output level is compressed or limited.
	These controls are used to set the output levels of channels 1 to 4, allowing to match the Dx38 to the input levels of the devices chained in sequence. Correctly setting these controls results in an improved S/N ratio. In most cases, good results are achieved when setting the controls to "-6". The digital output gain control should be used when higher output levels are needed. Use the controls OUT 1 - 4 to attenuate the output levels. It is not recommended to use the digital output gain control for massive attenuation, since this would decline the dynamic range of the D/ A-converters.
	Label of input or output channels.
	Clicking on the DSP button opens the Setup & Control window, which provides access to all DSP and speaker parameters.

DSP

The DSP pages provide overview and access to all DSP parameters of the sound system processor. Within this window you can use the Flow Diagram Selector to link to different function groups.

FLOW DIAGRAM SELECTOR

The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.

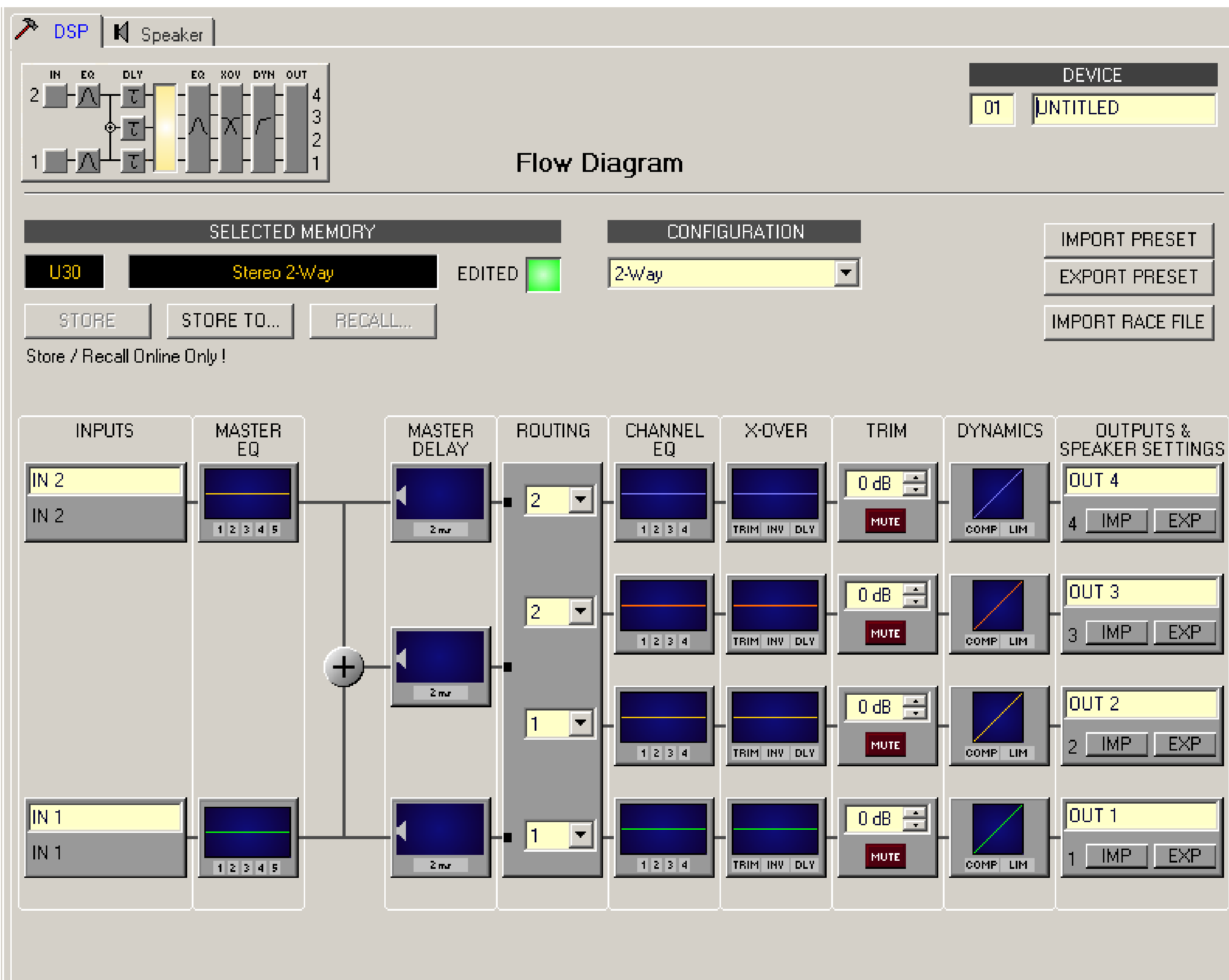


A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

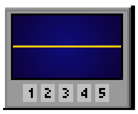
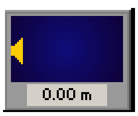
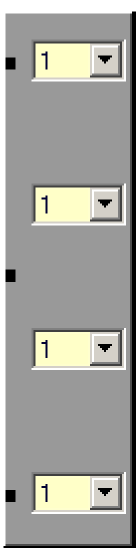
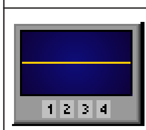
Element	**Description**
FLOW DIAGRAM	The signal flow display provides an overview of an amplifier's DSP settings. This area also includes all controls for the preset location/file management, configuration settings and RACE file import.
MASTER EQ	The MASTER EQ page provides access to the two 5-band parametric equalizers of the sound system processor inputs.

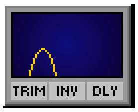
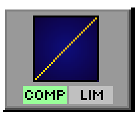
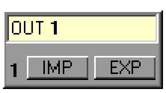
MASTER DELAY	This page allows the programming of delay lines for the channels A and B as well as for the summed input A+B.
CHANNEL EQ	The CHANNEL EQ page offers access to the 4-band parametric equalizers of the sound system processor outputs for speaker equalization.
X-OVER	Frequency crossover-filters as well as the parameters gain, polarity and alignment-delay for all output channels are located in the X-OVER area.
DYNAMICS	This page provides access to compressor and limiter of each channel.

The FLOW DIAGRAM window shows a signal flow diagram, which offers a quick overview of all DSP setting of the sound system processor. Labeling and routing channels, editing trim and mute can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for using presets, configurations and RACE files are also accessible from this window.
The FLOW DIAGRAM window opens when clicking on the first, fourth or eight block in the Flow Diagram Selector.



Function Blocks

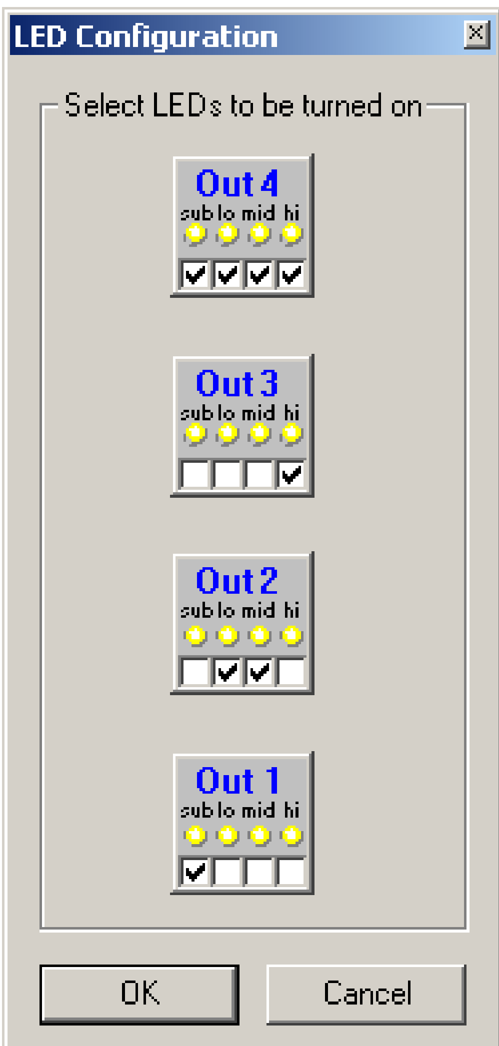
Element	Description
	Input Block: The text field allows specifying a name for the corresponding input channel. A click with the right mouse button onto IN 1 or IN 2 opens the Copy & Paste menu, which allows copying all parameters of the corresponding input channel (Master EQ, Master Delay) to any other input channel within the same project.
	Master EQ Block: The Master EQ block displays the 5 Master EQs of the corresponding input channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Master EQ block. A single click with the left mouse button onto this block opens the MASTER EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Master Delay Block: This displays the Master Delay of the input channels. The corresponding LED signals whether a delay has been programmed or not. The delay-value is displayed together with the measurement unit next to the LED. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the MASTER DELAY page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Master Delay block to any other Master Delay block within the same project.
	Routing Block: Here you can assign the output channel routing. The selection can be performed using the four combination boxes. A click with the right mouse but- ton onto routing block opens the Copy & Paste menu of all DSP settings, which allows copying all DSP parameters of an sound system processor to any other sound system processor within the same project.
	Channel EQ Block: The Channel EQ block displays the 4 Channel EQs of the corresponding output channel. The 4 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.

	X-Over Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X- Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse button onto this block opens the X-OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corres- ponding X-Over block to any other X-Over block within the same project.
	Level Block: The numerical field is identical to the numerical field below the level controls in the Userpanel. The MUTE button is for attenuating the output level of the corresponding output to –∞. Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is vir- tually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Dynamics Block: This block provides graphical display of the dynamics functions of the corresponding output. The two LEDs indicate whether compressor or limiter have been activated. The graph provides indication of the set values. A single click with the left mouse button onto this block opens the DYNAMICS page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Dynamics block to any other Dynamics block within the same project.
	Output Block: The text field allows specifying a name for the corresponding output channel. A click with the right mouse button onto OUT 1 to OUT 4 opens the Copy & Paste menu, which allows copying all parameters of the corresponding output channel (Routing, Channel EQ, X-Over, Dynamics) to any other output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied.

Status Indication

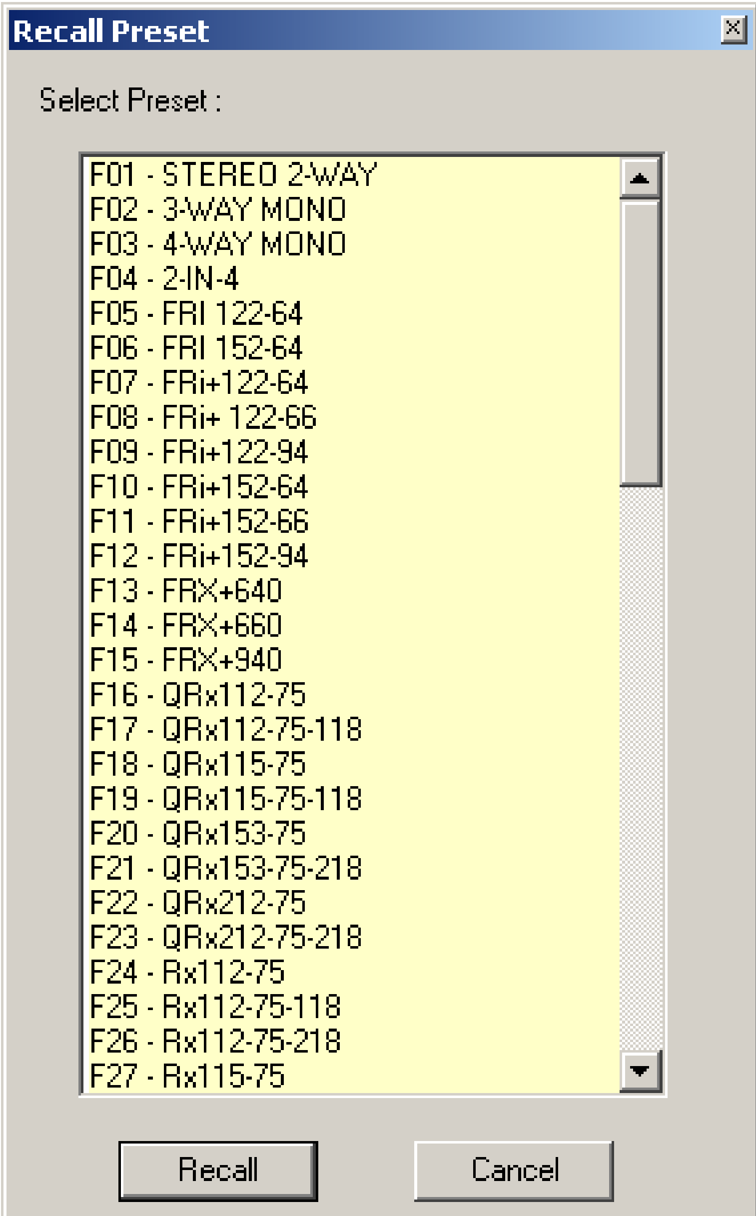
Element	Description
	Shows the number of the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	Indicates the name of the actually audible preset.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

4-Way	The Dx38 offers 7 pre-defined configurations. A configuration is a basic setting that includes the routing of inputs and outputs, the function of the outputs (Sub, Lo, Mid, Hi, Fullrange), as well as the kind and amount of parameters. If you do not want to use one of the pre-defined configurations, you have also the possibility to operate the Dx38 in Full Edit mode. This mode offers access to all parameters and basically any input/output routing can be programmed. Additionally, the output assignment (output function - Sub, Lo, Mid, Hi, Fullrange) is also freely definable.
LED CONFIG	If the sound system processor is operated in Full Edit Mode, clicking the button LED Config opens the LED Configuration dialog.



Recall a Presets

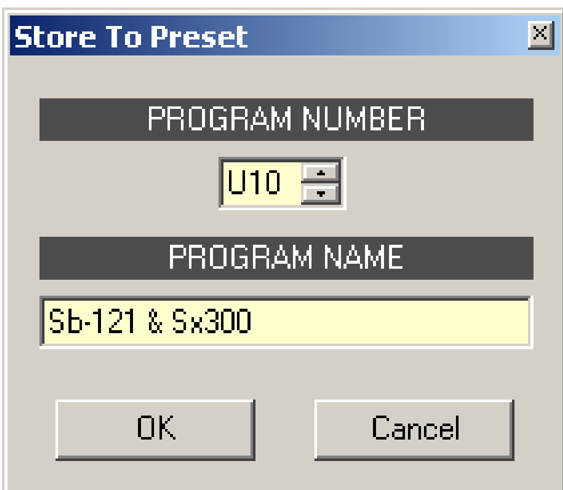
Element	Description
RECALL...	DClicking the button RECALL... opens the Recall Preset dialog, where you can select and recall a Preset.



Caution!
The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

Store a Presets

Element	Description
STORE	STORE saves all momentary set DSP parameters together with the entered name into the specified preset. Store is only possible if a user program number is selected.
STORE TO...	Clicking the STORE TO.... button opens the Store To Preset dialog. In this dialog the Program Number can be selected and the corresponding Program Name can be entered.



Import / Export a Preset

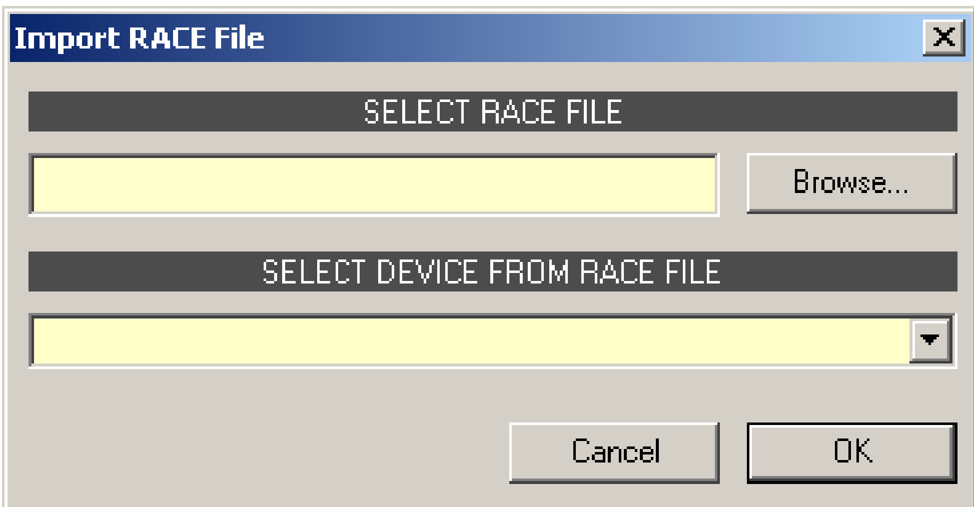
IRIS-Net allows the storing of all DSP parameters of an sound system processor together with the according preset name in a file, and to load sound system processor parameters from these files. Therefore, IRIS-Net creates a sub-directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory \Presets is permissible.

Element	Description
IMPORT PRESET	After clicking onto IMPORT PRESET appears an “Open File ...” dialog box. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a “Save File ...” dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store all DSP parameters together with the corresponding file name. ”.ds” is automatically added as file extension.

Import of EV RACE Files

IRIS-Net allows importing loudspeaker presets that have been created in Electro-Voice RACE.

Element	Description
	Clicking IMPORT RACE FILE opens the Import RACE File dialog.



First, you have to select the desired RACE file by use of the Browse... button. Because a RACE file can hold the data of up to 31 EV Dx38, you need to continue by selecting the desired device from the RACE file within the dialog ”SELECT DEVICE FROM RACE FILE”. Clicking onto ”OK” button completes the process.

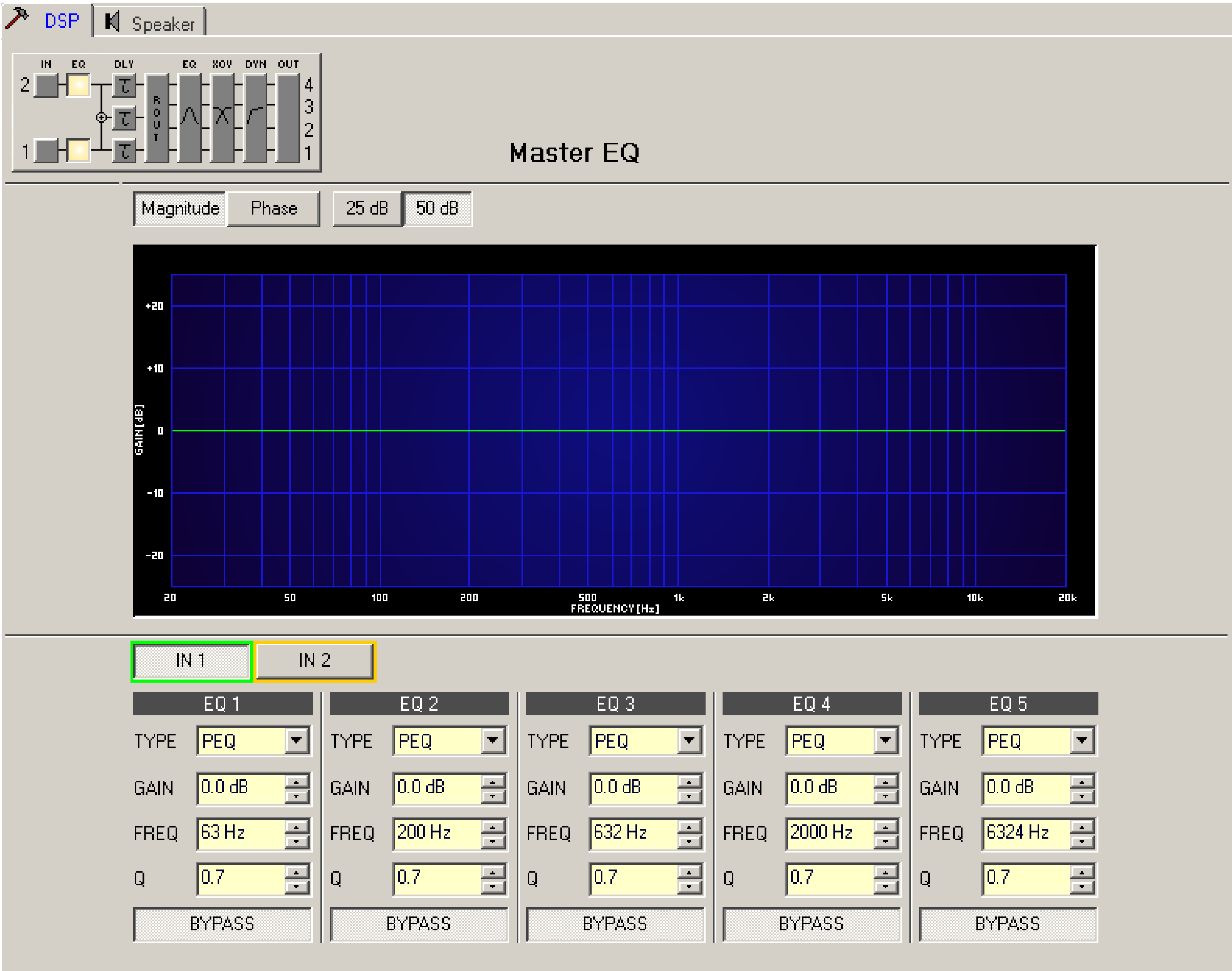
Caution!

The loaded RACE file becomes instantly audible when in on-line mode. Be sure to select the desired file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!

Consequences

MASTER EQ

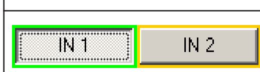
Both input channels of the sound system processor employ 5-band parametric equalizers each, which allow programming highly variable full-range speaker equalization to match a PA-system to different environmental and acoustical requirements. In many cases post-mixing console parametric equalization becomes redundant. The Master-EQ is selected by clicking on the second block of the flow diagram selector or on the MASTER EQ block in the full-scale flow diagram.



Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

Element	Description
	Switch for selecting input 1 (IN 1) or input 2 (IN 2) for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding out- put to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows conve- nient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE Hipass	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. – PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. – Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following edit- able parameters: frequency, slope and gain. – Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
SLOPE 12dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
FREQ 80 Hz	63 / 200 / 632 /	20 Hz...20	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving
	2000 / 6324 Hz	kHz	and Hi / Lo pass filters.
Q +1.0	0.4	0.4...20.0	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		(PEQ),	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the res-
		0.4...2.0 (Hi-/	response of Hi, Lo and All pass filters with slopes of 12dB/oct.
		Lopass)	
GAIN +2.5 dB	0 dB	-12...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

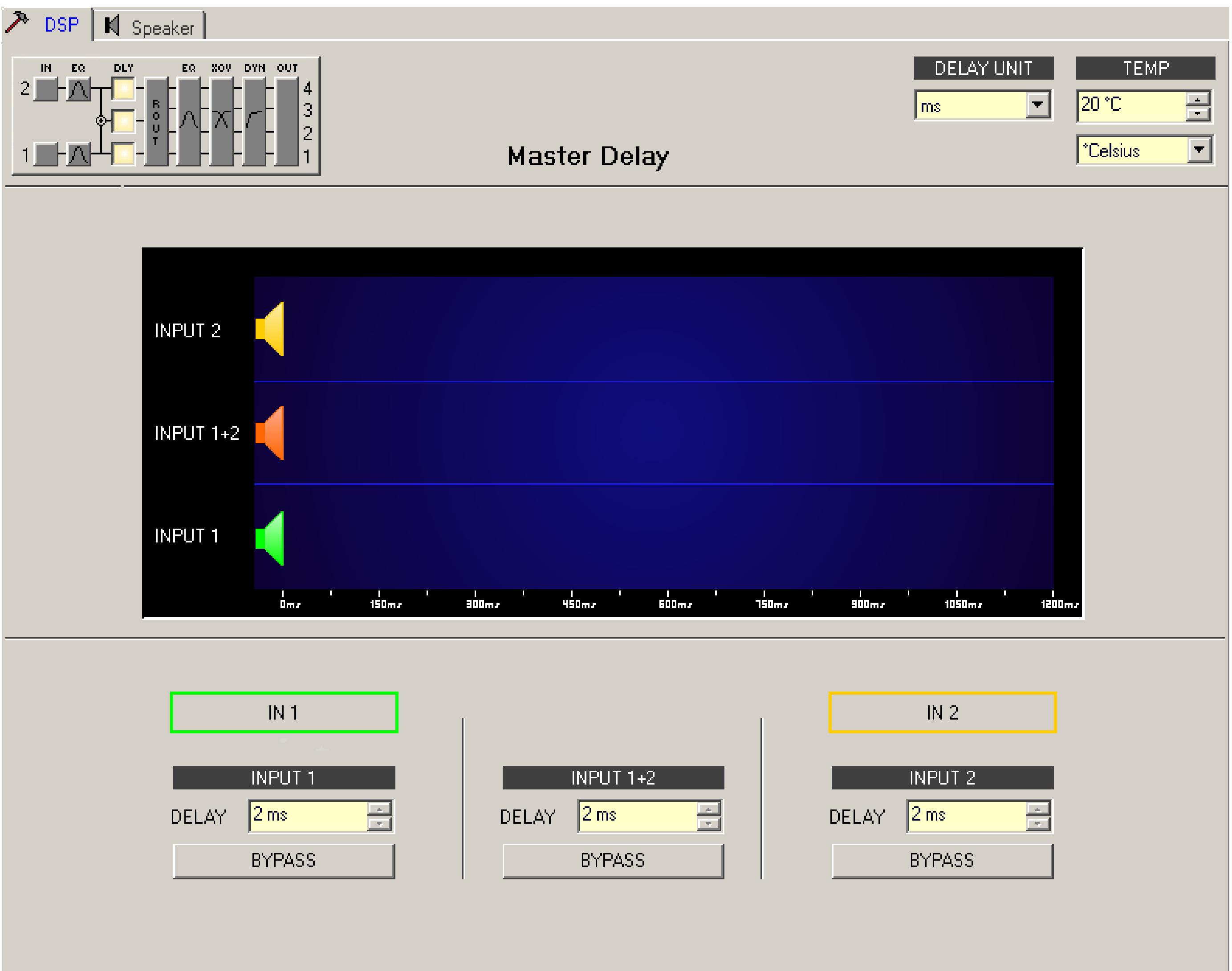
Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its amplification (depending on the selected filter type) by

moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

MASTER DELAY

Individual master delays can be set for each input channel of a remote amplifier. Setting a different delay for the summed signal of the two input channels is also possible. Master Delays are mainly used to compensate for different natural delay times in the audio signal, as they are common when two sound sources reproducing identical audio information are located further apart.
You can select the master delay window by clicking onto the third block in the Flow Diagram Selector or onto the MASTER DELAY block in the flow diagram.



Channel Parameters

Element	Default	Range	Description
IN 1			Channel name
INPUT 1			Channel identification
DELAY 35 m	2.0 ms	2...900 ms	DELAY allows delaying the corresponding input channel's audio signal by an adjustable period of time.
BYPASS			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

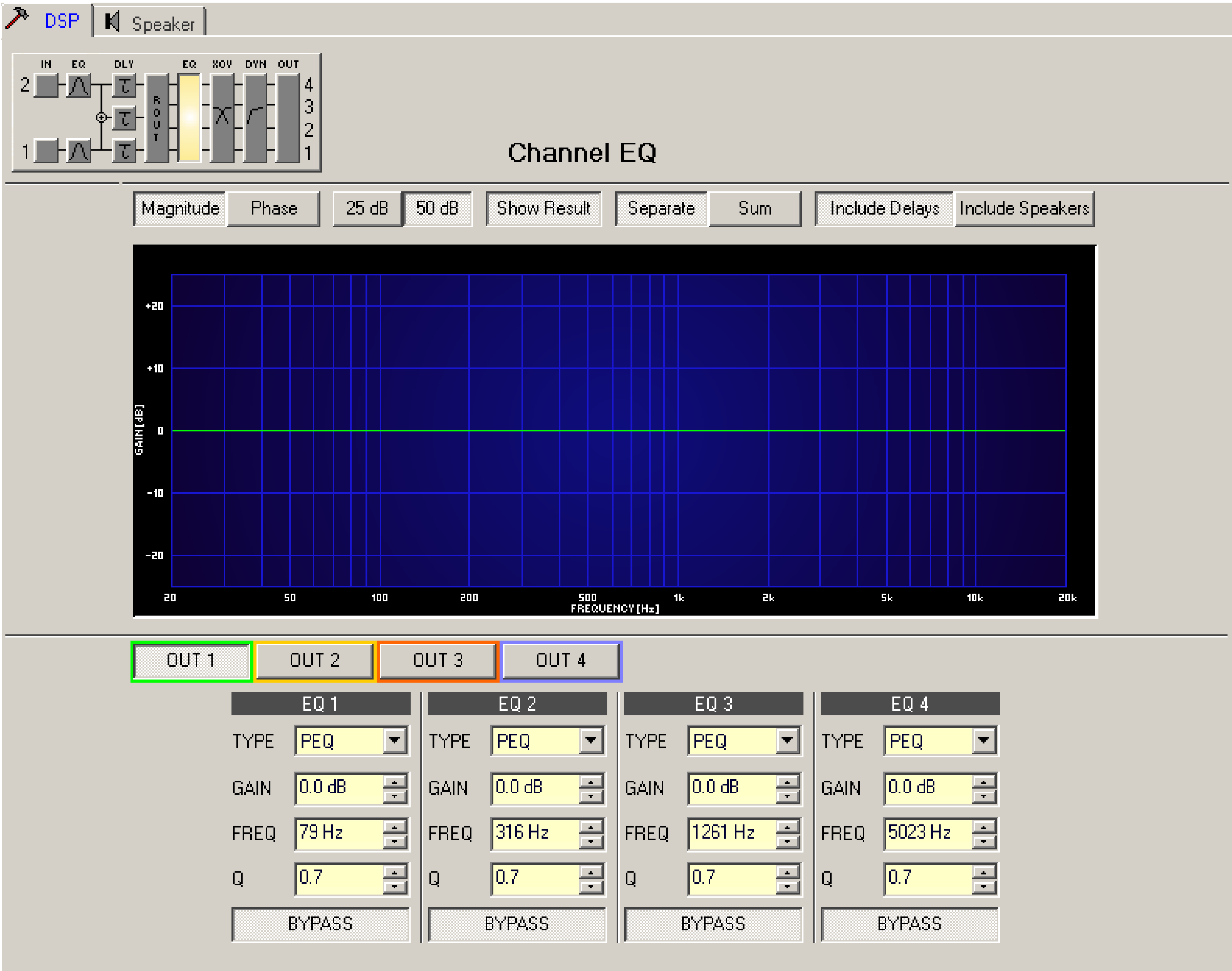
General Parameters

Element	Default	Range	Description
DELAY UNIT m	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
TEMPERATURE +23 °C °Celsius	20 °C	-20...60 °C -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by "Dragging the Mouse" in the Graphics Display
The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel's delay time. A delay's title "lights" in color as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

CHANNEL EQ

All output channels of the sound system processor employ 4-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select "All pass" as filter type, these filters are identical to the ones of the Master-EQ's.
The Channel-EQ is selected by clicking on the fifth block of the flow diagram selector or by on the CHANNEL EQ block in the full-scale flow diagram.



Graphics Display Indication

Element	**Description**
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the sound system processor out- puts. The audible result is displayed in bright colors while “electrical” graphs are indicated in dark colors.
Separate Sum	Selecting ”separate” results in a separated display of the sound system processor channels’ transfer functions while ”sum” shows the summed signal of the sound system processor channels.

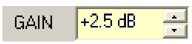
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signal reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the "Speaker" register sheet.

Channel Selection

Element	Description
OUT 1	Switch for selecting output 1, 2, 3 or output 4 for filter editing. A click with the right mouse button opens the "Copy & Paste" menu, which allows convenient copying all EQs of the corresponding output to any other EQ-filter bank within the same project.

Filter Parameters

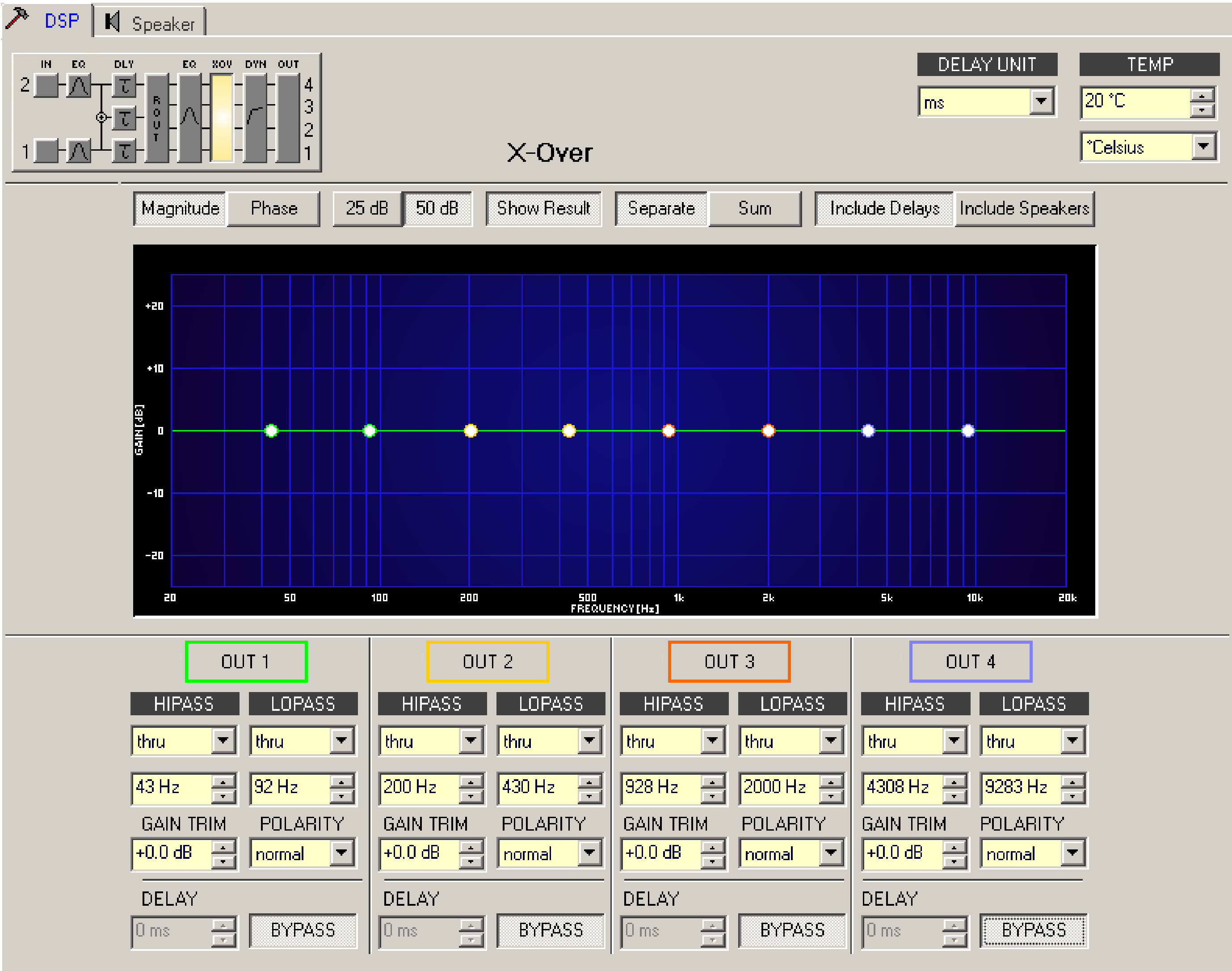
Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the "Copy & Paste" menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE Hipass	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. – PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. – Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. – Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. – Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 12dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
FREQ 80 Hz	79 / 316 / 1261 / 5023Hz	20 Hz...20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q +1.0	0.7	0.4...40.0 (PEQ),	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		0.4...2.0 (Hi-/	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the

		Lopass), 0.4...2.0 (All-pass)	response of Hi, Lo and All pass filters with slopes of 12dB/oct.
	0 dB	-18...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with All pass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via “Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

X-OVER

The X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch, which are provided for each output channel of a sound system processor. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.
Clicking on the sixth block in the Flow Diagram Selector or on the X-OVER block in the large signal flow diagram opens the X-Over window.



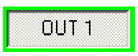
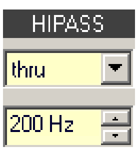
Graphics Display Indication

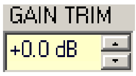
The graphics display offers several different indication modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (Master EQ, Channel EQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Result	Displays the resulting transfer function of all filter and level trim settings and therefore the visible respectively audible result at the sound system processor outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.
Separate Sum	The ”Separate” switch allows separate indication of the two amplifier channels’ transfer functions. The ”Sum” switch causes display of the summed signal of the two channels.

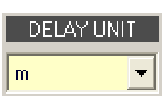
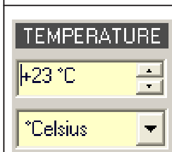
	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the "Speaker" register sheet.

Channel Parameters

Element	Default	Range	Description
			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all X-Over parameters of the corresponding output to any other X-Over within the same project.
	thru, 43 / 200 / 928 / 4308 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI-PASS filters within the same project.
	thru, 92 / 430 / 2000 / 9283 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.

	0 dB	-30 dB...6 dB	GAIN TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
	normal	normal, inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to”Sum”).
	0.0 ms	0.0...900.0 ms	DELAY allows delaying the audio signal of the corresponding output by an adjustable period of time. This delay method is typically used as time-alignment-delay to overcome nega- tive sound effects like they result from different distances between loudspeaker systems within one cabinet or the positioning of speakers in a PA-installation that otherwise would cause a high amount of natural delay.
			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

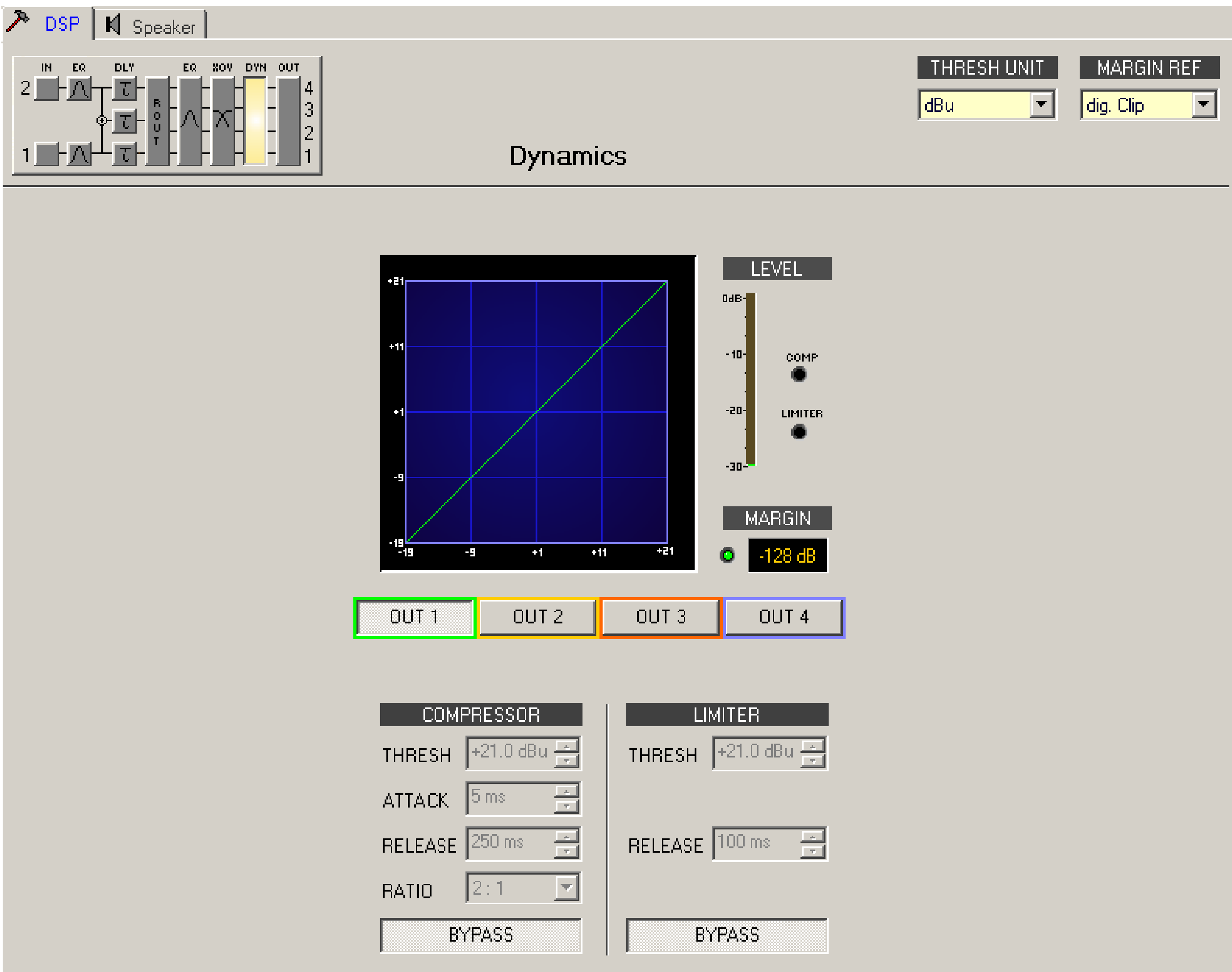
Element	Default	Range	Description
	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
	20 °C	-20...60 °C -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing X-Over Filters by “Dragging the Mouse” in the Graphics Display

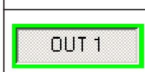
Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right. A filter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling. An extra white graph is displayed in addition, representing the frequency response of the corresponding selected filter.

DYNAMICS

Each output channel of the sound system processor offers a compressor and a limiter. These functions can be accessed via the dynamics window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.
Clicking onto the seventh block in the Flow Diagram Selector or double clicking onto the DYNAMICS block in the large flow diagram opens the dynamics window.



Channel Parameters

Element	Description
	Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all dynamics parameters of the corresponding channel to any other channels within the same project.

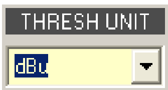
Compressor Parameters

Element	Default	Range	Description
COMPRESSOR			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all compressor parameters of the corresponding channel to any other channels within the same project.
THRESH +55.7 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	The THRESHOLD parameter determines the audio signal level above which the compressor starts operating.
ATTACK 5 ms	5 ms	0...99 ms	ATTACK determines how fast the compressor reduces amplification when the threshold is exceeded.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the compressor returns to normal amplification, after the audio signal level declined the threshold.
RATIO 4 : 1	2:1	1: 1, 1.4: 1, 2: 1, 4: 1, 8:1	RATIO determines the amount of compression that the audio signal is compressed when exceeding the threshold. The setting of 4 : 1 for example relates to a reduction of the audio signal by the factor 4.
BYPASS			BYPASS switches the compressor on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the compressed and non-compressed audio signals.

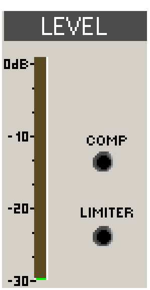
Limiter Parameters

Element	Default	Range	Description
PEAK LIMITER			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all limiter parameters of the corresponding channel to any other channels within the same project.
THRESH +55.7 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	THRESHOLD determines the audio signal level above which the limiter starts operating.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
BYPASS			BYPASS switches the limiter on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the limited and non-limited audio signals.

General Parameters

Element	Default	Range	Description
	dBu	dBu / Volts	This lets you select the unit for the threshold parameter. The selected setting applies to compressor and limiter as well.
	dig. Clip	dig. Clip, Limiter Thresh	This lets you set the absolute level for margin indication. You can select between ”Digital Clip” (relates to +21 dBu) and ”Limiter Threshold”. The margin level indicates the distance between signal level and the set absolute level. The displayed mar- gin always relates to the highest actual signal level reading.

Indications

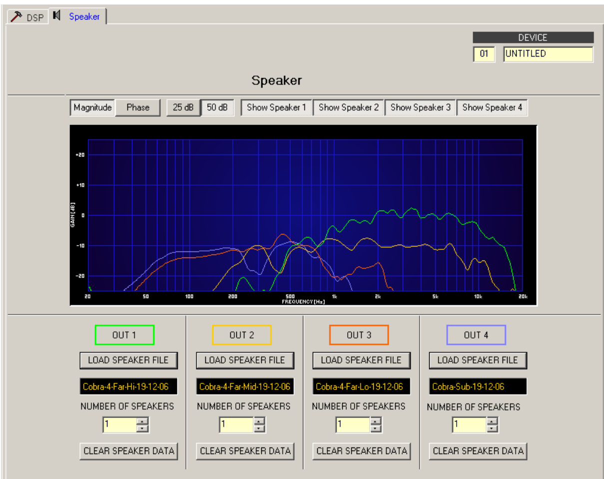
Element	Description
	These indicator shows the reduction in dB that is applied to the audio signal by the compressor (COMP) or limiter. Level reduction is indicated as vertical yellow bar graph.
	The margin level indicates the distance between signal level and the set absolute level. The displayed margin relates to the highest actual signal level reading since the last reset of the indicator. The LED changes from green to red as soon as the signal level reaches or exceeds the set absolute level (Digital Clip / Limiter Threshold). A click with the right mouse button onto the margin level followed by click onto Reset re-sets indication.

Editing Compressor / Limiter Parameters by “Dragging the Mouse” in the Graphics Display
Active compressors or limiters (bypass button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding compressor or limiter by vertically dragging the mouse. A click with the right mouse button onto the white dot of a compressor and keeping the mouse button pressed down lets you edit the ratio of compression.
A compressor’s / limiter’s title “lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

Speaker

The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the sound system processor channels and display the acoustic results. The speaker system datasets, which are provided as “speaker files” (*.spk), contain factory-measured frequency- and phase responses of loudspeaker systems. The speaker data as well as any settings made in this window have no direct influence on the transfer function of the sound system processor. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher

quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of sound system processor plus speaker transfer functions.
Clicking on the Speaker tab in the Setup & Control window opens the Speaker page.



Indication on the Graphic Display

Element	Description
Magnitude Phase	Switch for toggling between frequency response (magnitude) and phase response (phase) display
25 dB 50 dB	Switch for adjusting the scale of the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Speaker 1	Switching the display of the corresponding speaker data for an sound system processor channel on/off is performed using the ”Show Speaker 1” to ”Show Speaker 4” switches.

Channel Parameters

Element	Default	Range	Description
OUT 1			Channel name.
LOAD SPEAKER FILE			Clicking the button LOAD SPEAKER FILE opens a dialog that allows the selection of the desired speaker file.
No Speaker			The name of the loaded loudspeaker model is shown in the black-shaded field.

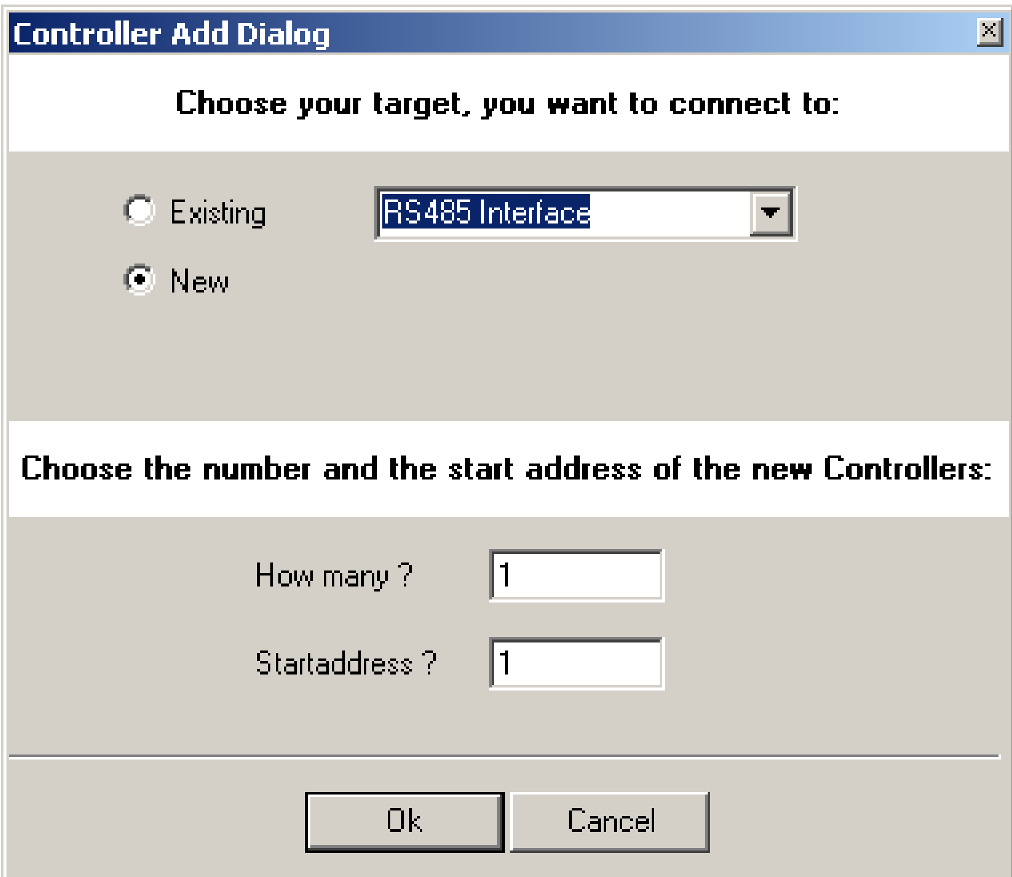
	1	1...8	The NUMBER OF SPEAKERS parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel. Setting an amount from 1 to 8 is possible.
			Clicking the CLEAR SPEAKER DATA button clears the previously loaded measured speaker data of the selected channel.

DSP 244

The DYNACORD DSP 244 is a universal Digital Sound System Processor that provides 2 inputs and 4 outputs; plus internal summing of the inputs 1 and 2. Via matrix it is possible to assign the outputs to any input or to the sum of the inputs. It is further possible to establish the following configurations: Stereo or Dual 2-Way systems, 3-Way + Direct and 4-Way systems, each with Mono Sub-channel, but also full range systems.
High and low-pass filters are provided for the frequency crossover functions in all operation modes. The selection includes Linkwitz-Riley, Butterworth and Bessel type filters with switchable slopes between 6, 12, 18 and 24 dB/oct. A huge number of additional filters offer extremely flexible correction of the frequency response. Each input incorporates a 5-band equalizer, allowing assigning high and low-pass, high and low-shelving or parametric peak-dip filters to its individual filter sections. Next to the frequency crossover filters, four additional filters are employed in each output channel, which also can be set to work as high or low-pass, high or low-shelving filters, parametric peak-dip filters, or all-pass filters. Additional filtering is provided through 2. Order high-passes for the realization of B-6 alignment, or special LPN filters (Low-Pass Notch filters) for correcting the frequency and phase responses of optimally vented woofer cabinets. Each channel additionally provides a delay, a polarity switch, a programmable level control and a digital compressor / limiter while the master delays are located in the input channels.
The user can choose between two operation modes: the ”No Edit Mode” allows to simply select the required combination of loudspeaker systems from the factory preset program list. Afterwards, the appliance is optimally matched to the sound system and can be operated instantly. The ”Full Edit Mode” on the other hand offers access to all parameters, allowing to freely program and store basically any setting. A total number of 80 memory addresses - 50 preset and 30 freely assignable user-programs - are available.
Within the DSP 244, AD/DA conversion is taken care of by linear 24-bit converters; where the AD-section employs 128 times oversampling, gain-ranging Sigma-Delta converters. The DA-section offers 128 times oversampling Sigma-Delta converters. The overall signal processing is performed by two 24-bit Motorola signal processors.
Additional features are:
– FLASH memory for software and preset updates via serial interfaces
– PC-based operation and configuration software IRIS-Net
– Standard MIDI-interface and RS-232 interface
– RS-485 interface or switching contacts optionally available
– Back-lit graphic-display with 122 x 32 dots
– Inputs and outputs are electronically balanced, XLR-type connectors
– Input transformer-balancing is optionally available
– Input / Output level controls, Output-Mute switch, channel function indicators SUB, LO, MID, HI
– Input / Output meter instruments, compressor and clipping LEDs

DSP 244 Device

Start by creating an DSP 244 Device in your IRIS-Net project. Drag an DSP 244 from the Object Bar’s Devices category or from the Devices window into the worksheet. The following dialog box appears:

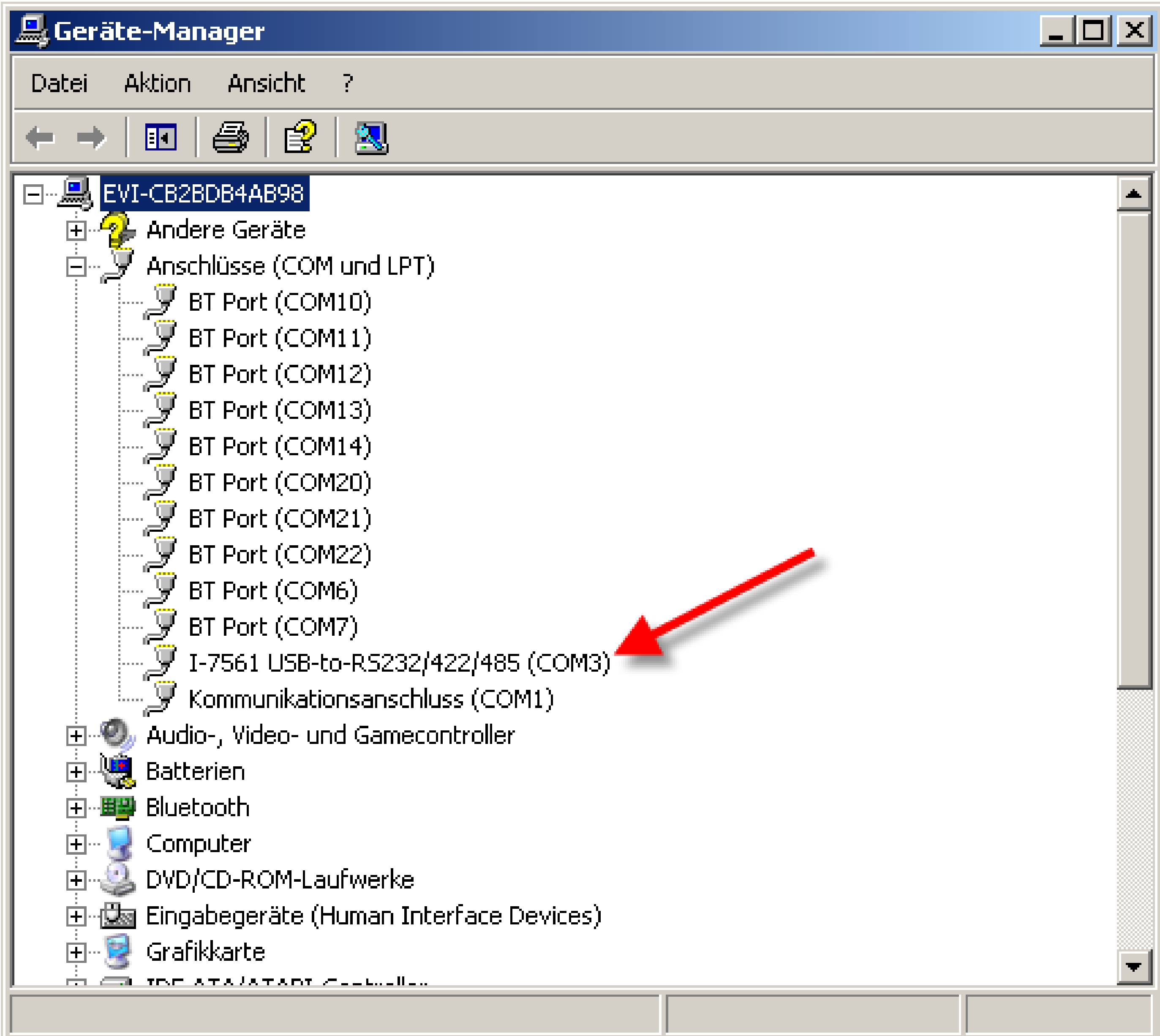


Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of DSP 244 Devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the Ctrl key and click on the device. In either case a successfully selected device is shown with a red border around it.

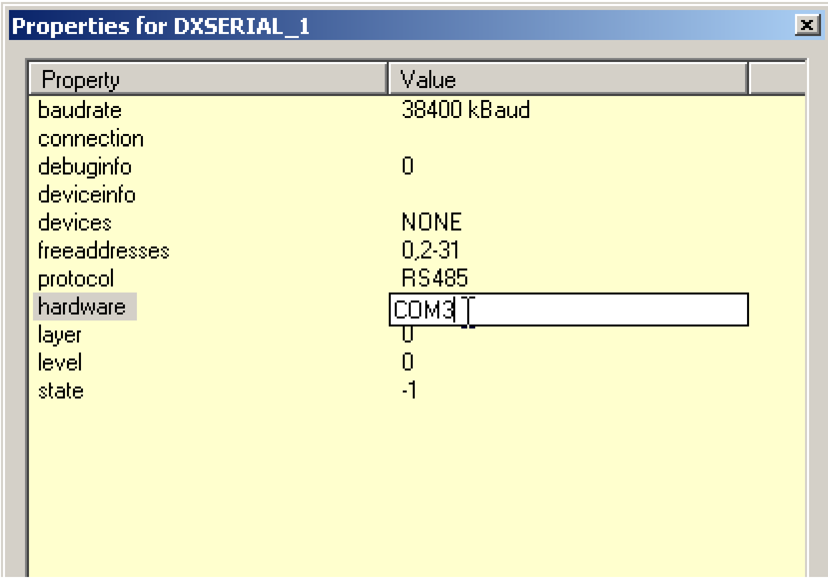
Double clicking on an DSP 244 device icon opens the Userpanel.

CONFIGURATION OF THE INTERFACE

For using the DSP 244 device in IRIS-Net the used COM port of the PC has to be selected. When using an USB-RS-485 adapters the system internal COM port can be found in the Control Panel, see following picture.



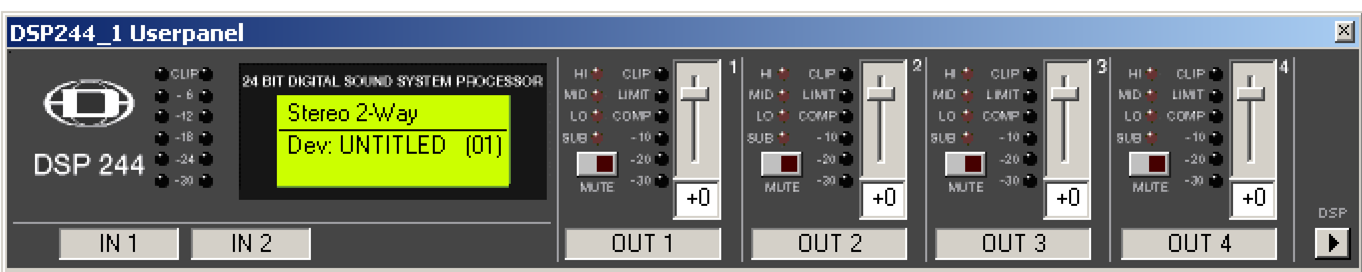
In this example COM port 3 is used by the adapter. The value „COM3" must be entered for the Property „hardware" of the serial interface in IRIS-Net.



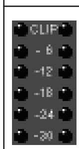
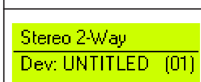
Reference

DSP 244 USERPANEL

The DSP 244 Userpanel provides access to the controls and indications at the DSP 244 front panel. The complete functionality available at the DSP 244 LC-Display can be accessed via pressing the DSP button.



Indications and Functions of the DSP 244 Userpanel

Element	Description
	The level meter instruments are meant for optical monitoring of the input signal levels, individually showing the peak value of the correspondent input signal. The input control should be set to a position so that the meter instruments indicate a level between -6 and -12 dB. To prevent inter- nal clipping, make sure that the CLIP LEDs are not lit.
	Line 1 displays the description of the selected User Memory. Line 2 displays the description of the device and its address at the RS-485 bus.
	These LEDs indicate, which frequency band the corresponding channel is set to. If a channel is configured for full range operation, all its func- tion-LEDs are simultaneously lit.
	These keys allow to mute the output signal of the corresponding output channels. Pressing a key once switches the mute-function ON; the key’s red LED lights. Pressing the button again switches the mute-function OFF; the key’s LED is dimmed again.

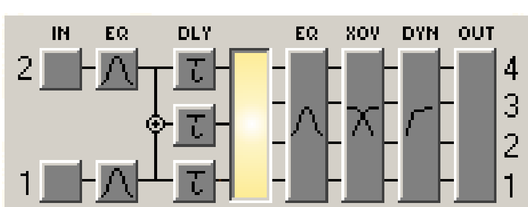
	These LEDs indicate the peak level of the corresponding outputs. The Dx38 should be operated in a range, so that the clip-LEDs are not lit. Otherwise, this could lead to internal clipping. The COMP/ LIMIT-LEDs light, when the compressor / limiter of the corresponding channel is activated; e. g.: when the audio signal level exceeded the previously set threshold and therefore the output level is compressed or limited.
	These controls are used to set the output levels of channels 1 to 4, allowing to match the Dx38 to the input levels of the devices chained in sequence. Correctly setting these controls results in an improved S/N ratio. In most cases, good results are achieved when setting the controls to "-6". The digital output gain control should be used when higher output levels are needed. Use the controls OUT 1 - 4 to attenuate the output levels. It is not recommended to use the digital output gain control for massive attenuation, since this would decline the dynamic range of the D/ A-converters.
	Label of input or output channels.
	Clicking on the DSP button opens the Setup & Control window, which provides access to all DSP and speaker parameters.

DSP

The DSP pages provide overview and access to all DSP parameters of the sound system processor. Within this window you can use the Flow Diagram Selector to link to different function groups.

FLOW DIAGRAM SELECTOR

The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a yellow engaged field.

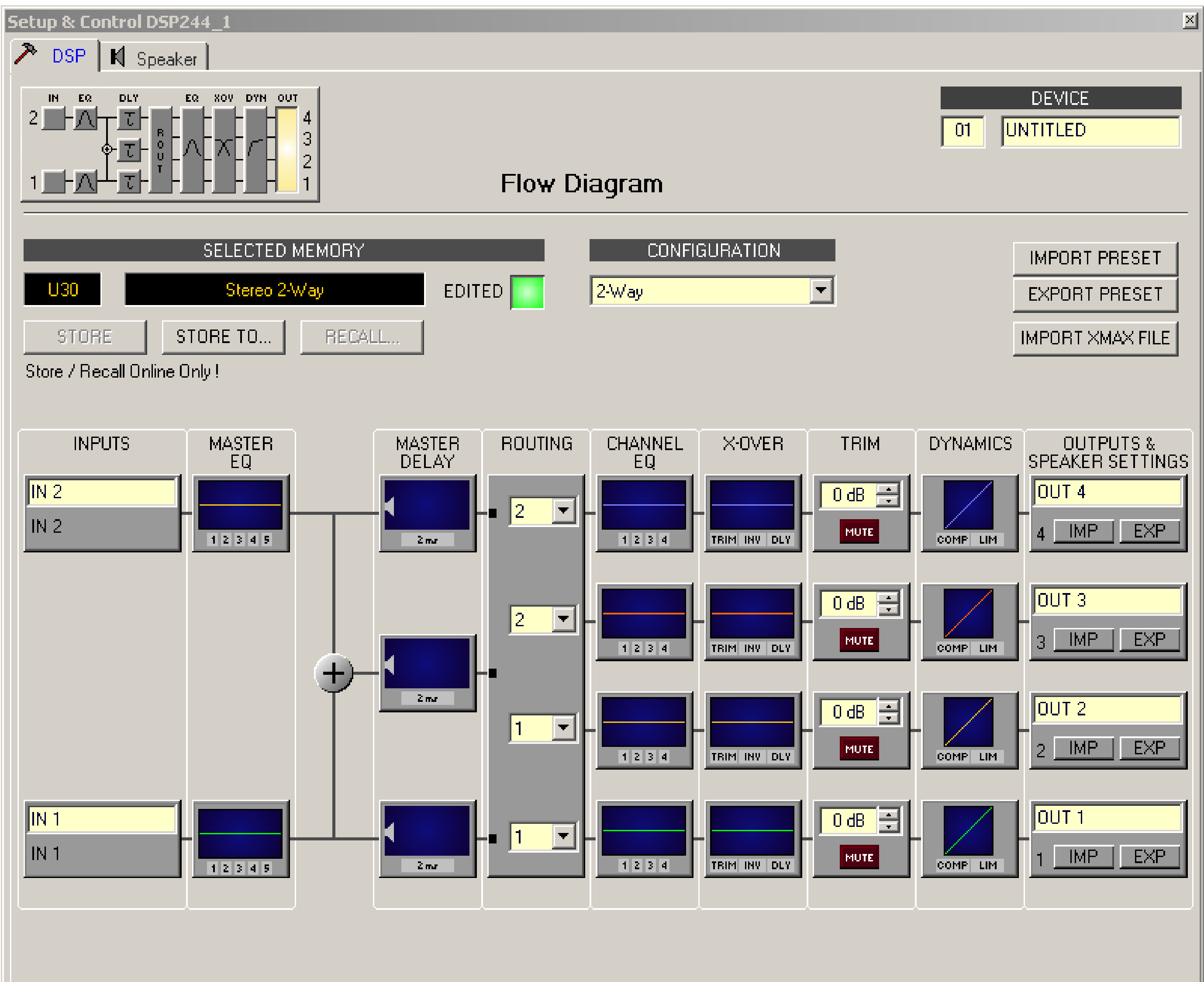


A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

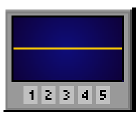
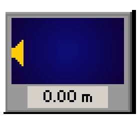
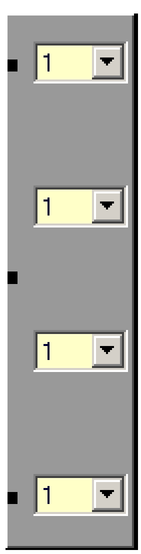
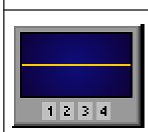
Element	**Description**
FLOW DIAGRAM	The signal flow display provides an overview of an amplifier's DSP settings. This area also includes all controls for the preset location/file management, configuration settings and RACE file import.
MASTER EQ	The MASTER EQ page provides access to the two 5-band parametric equalizers of the sound system processor inputs.
MASTER DELAY	This page allows the programming of delay lines for the channels A and B as well as for the summed input A+B.
CHANNEL EQ	The CHANNEL EQ page offers access to the 4-band parametric equalizers of the sound system processor outputs for speaker equalization.

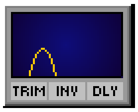
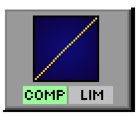
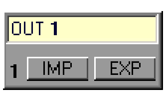
X-OVER	Frequency crossover-filters as well as the parameters gain, polarity and alignment-delay for all output channels are located in the X-OVER area.
DYNAMICS	This page provides access to compressor and limiter of each channel.

The FLOW DIAGRAM window shows a signal flow diagram, which offers a quick overview of all DSP setting of the sound system processor. Labeling and routing channels, editing trim and mute can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for using presets, configurations and RACE files are also accessible from this window.
The FLOW DIAGRAM window opens when clicking on the first, fourth or eight block in the Flow Diagram Selector.



Function Blocks

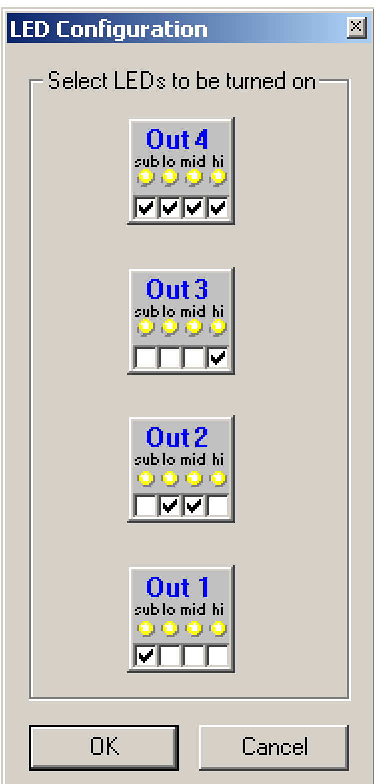
Element	Description
	Input Block: The text field allows specifying a name for the corresponding input channel. A click with the right mouse button onto IN 1 or IN 2 opens the Copy & Paste menu, which allows copying all parameters of the corresponding input channel (Master EQ, Master Delay) to any other input channel within the same project.
	Master EQ Block: The Master EQ block displays the 5 Master EQs of the corresponding input channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Master EQ block. A single click with the left mouse button onto this block opens the MASTER EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.
	Master Delay Block: This displays the Master Delay of the input channels. The corresponding LED signals whether a delay has been programmed or not. The delay-value is displayed together with the measurement unit next to the LED. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the MASTER DELAY page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Master Delay block to any other Master Delay block within the same project.
	Routing Block: Here you can assign the output channel routing. The selection can be performed using the four combination boxes. A click with the right mouse but- ton onto routing block opens the Copy & Paste menu of all DSP settings, which allows copying all DSP parameters of an sound system processor to any other sound system processor within the same project.
	Channel EQ Block: The Channel EQ block displays the 4 Channel EQs of the corresponding output channel. The 4 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the Channel EQ block. A single click with the left mouse button onto this block opens the CHANNEL EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other EQ block within the same project.

	X-Over Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X- Over parameters. Three additional LEDs indicate the status of gain trim, polarity and delay. A single click with the left mouse button onto this block opens the X-OVER page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other X-Over block within the same project.
	Level Block: The numerical field is identical to the numerical field below the level controls in the Userpanel. The MUTE button is for attenuating the output level of the corresponding output to −∞. Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Dynamics Block: This block provides graphical display of the dynamics functions of the corresponding output. The two LEDs indicate whether compressor or limiter have been activated. The graph provides indication of the set values. A single click with the left mouse button onto this block opens the DYNAMICS page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Dynamics block to any other Dynamics block within the same project.
	Output Block: The text field allows specifying a name for the corresponding output channel. A click with the right mouse button onto OUT 1 to OUT 4 opens the Copy & Paste menu, which allows copying all parameters of the corresponding output channel (Routing, Channel EQ, X-Over, Dynamics) to any other output channel within the same project. However, it is important to bear in mind that the DSP data only is being copied. Impedance or speaker data is not copied.

Status Indication

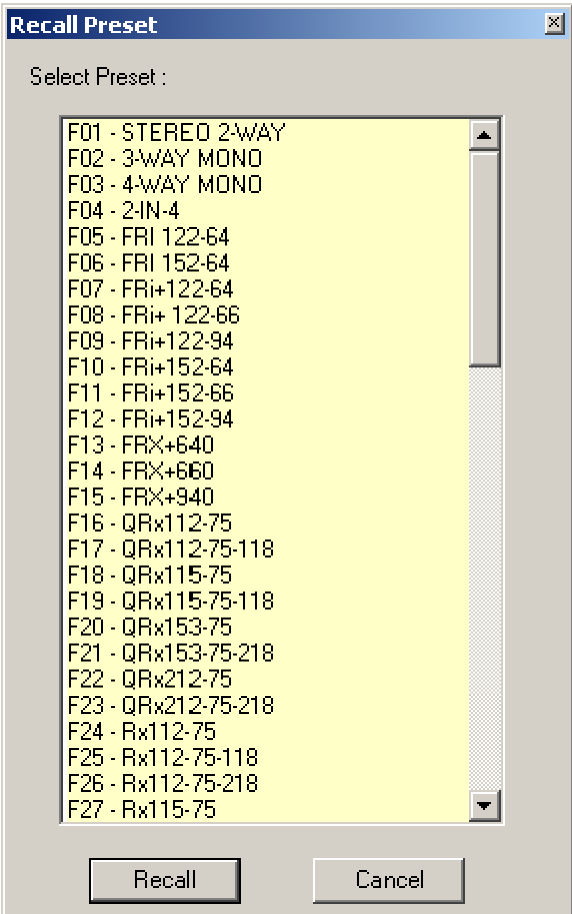
Element	**Description**
	Shows the number of the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	Indicates the name of the actually audible preset.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

	The DSP 244 offers 7 pre-defined configurations. A configuration is a basic setting that includes the routing of inputs and out- puts, the function of the outputs (Sub, Lo, Mid, Hi, Fullrange), as well as the kind and amount of parameters. If you do not want to use one of the pre-defined configurations, you have also the possibility to operate the DSP 244 in Full Edit mode. This mode offers access to all parameters and basically any input/output routing can be programmed. Additionally, the output assignment (output function - Sub, Lo, Mid, Hi, Fullrange) is also freely definable.
	If the sound system processor is operated in Full Edit Mode, clicking the button LED Config opens the LED Configuration dialog.



Recall a Presets

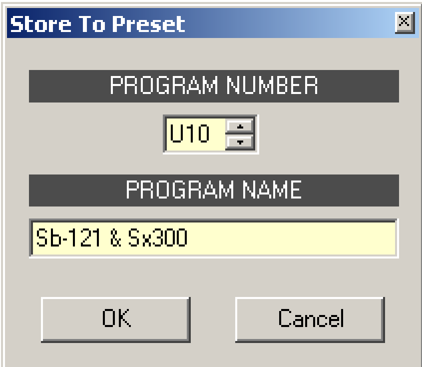
Element	Description
	Clicking the button RECALL... opens the Recall Preset dialog, where you can select and recall a Preset.



Caution!
The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

Store a Presets

Element	Description
STORE	STORE saves all momentary set DSP parameters together with the entered name into the specified preset. Store is only possible if a user program number is selected.
STORE TO...	Clicking the STORE TO.... button opens the Store To Preset dialog. In this dialog the Program Number can be selected and the corresponding Program Name can be entered.



Import / Export a Preset

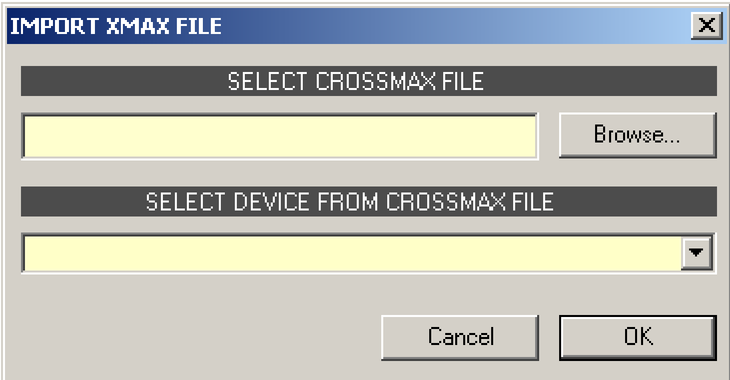
IRIS-Net allows the storing of all DSP parameters of an sound system processor together with the according preset name in a file, and to load sound system processor parameters from these files. Therefore, IRIS-Net creates a sub-directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory \Presets is permissible.

Element	Description
IMPORT PRESET	After clicking onto IMPORT PRESET appears an "Open File ..." dialog box. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a "Save File ..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click onto the SAVE button to store all DSP parameters together with the corresponding file name. ".ds" is automatically added as file extension.

Import of CrossMax Files

IRIS-Net allows importing loudspeaker presets that have been created in CrossMax.

Element	Description
IMPORT RACE FILE	Clicking IMPORT CrossMax FILE opens the Import CrossMax File dialog.

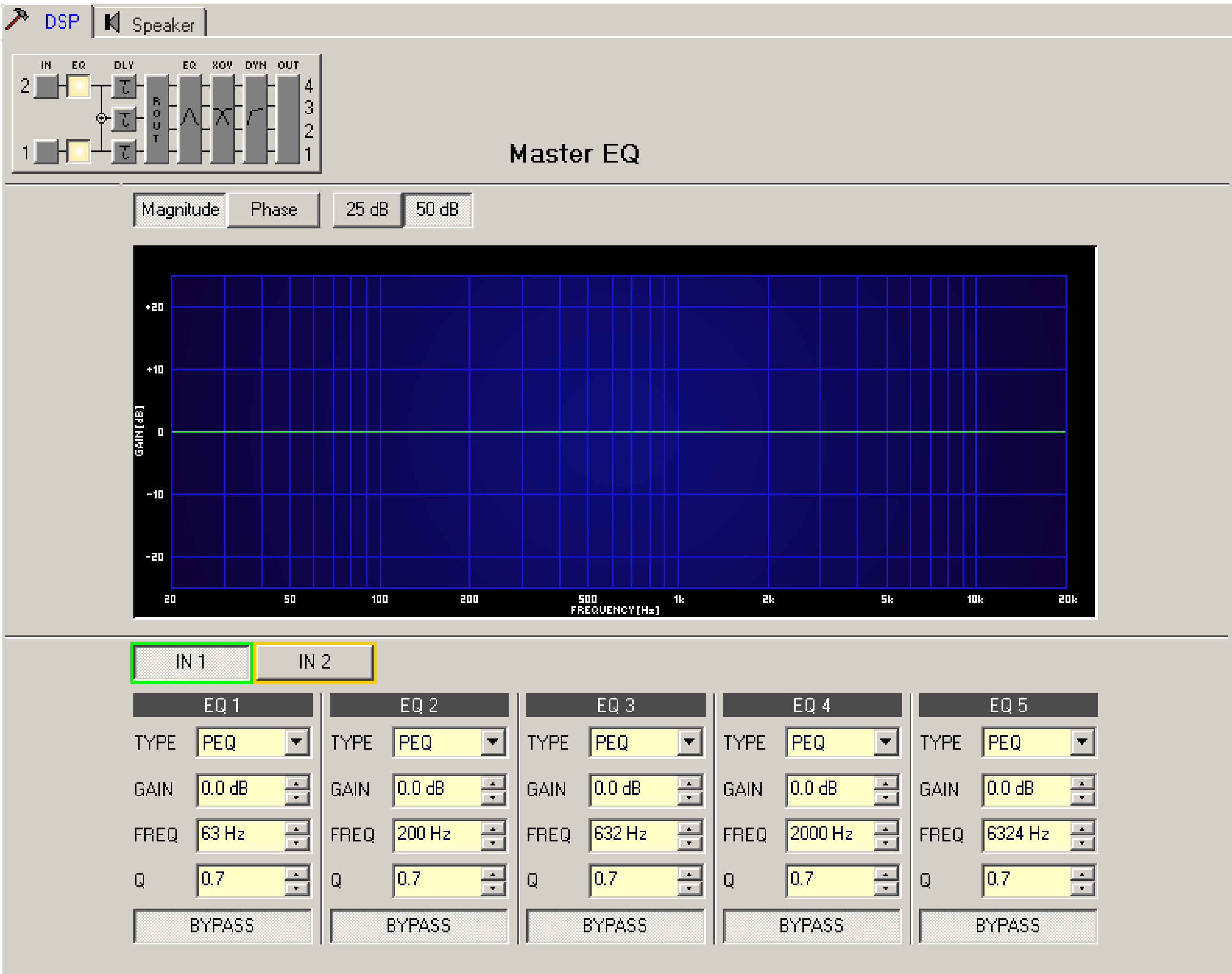


First, you have to select the desired CrossMax file by use of the Browse... button. Because a CrossMax file can hold the data of up to 31 DSP 244, you need to continue by selecting the desired device from the CrossMax file within the dialog ”SELECT DEVICE FROM CrossMax FILE”. Clicking onto ”OK” button completes the process.

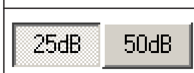
Caution!
The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

MASTER EQ

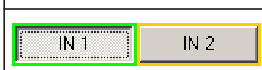
Both input channels of the sound system processor employ 5-band parametric equalizers each, which allow programming highly variable full-range speaker equalization to match a PA-system to different environmental and acoustical requirements. In many cases post-mixing console parametric equalization becomes redundant. The Master-EQ is selected by clicking on the second block of the flow diagram selector or on the MASTER EQ block in the full-scale flow diagram.



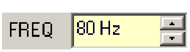
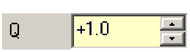
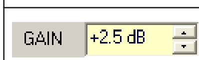
Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

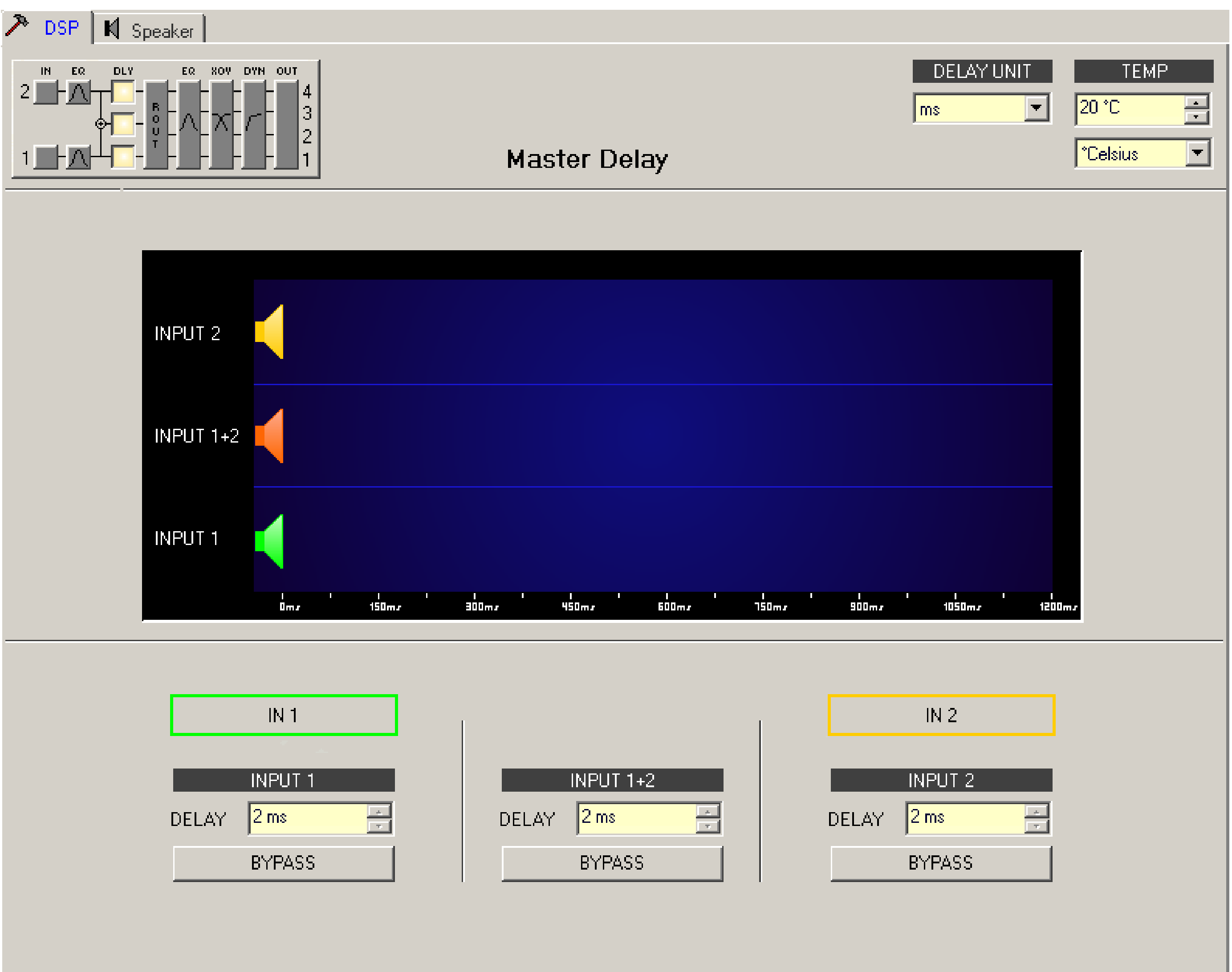
Element	Description
	Switch for selecting input 1 (IN 1) or input 2 (IN 2) for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding out- put to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows conve- nient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following edit- able parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
	63 / 200 / 632 /	20 Hz...20	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving
	2000 / 6324 Hz	kHz	and Hi / Lo pass filters.
	0.4	0.4...20.0	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		(PEQ),	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the res-
		0.4...2.0 (Hi-/	ponse of Hi, Lo and All pass filters with slopes of 12dB/oct.
		Lopass)	
	0 dB	-12...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via “Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its amplification (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

MASTER DELAY
Individual master delays can be set for each input channel of a remote amplifier. Setting a different delay for the summed signal of the two input channels is also possible. Master Delays are mainly used to compensate for different natural delay times in the audio signal, as they are common when two sound sources reproducing identical audio information are located further apart.
You can select the master delay window by clicking onto the third block in the Flow Diagram Selector or onto the MASTER DELAY block in the flow diagram.



Channel Parameters

Element	Default	Range	Description
IN 1			Channel name
INPUT 1			Channel identification
DELAY 35 m	2.0 ms	2...900 ms	DELAY allows delaying the corresponding input channel’s audio signal by an adjustable period of time.
BYPASS			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

Element	Default	Range	Description
DELAY UNIT m	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
TEMPERATURE +23 °C °Celsius	20 °C	-20...60 °C -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

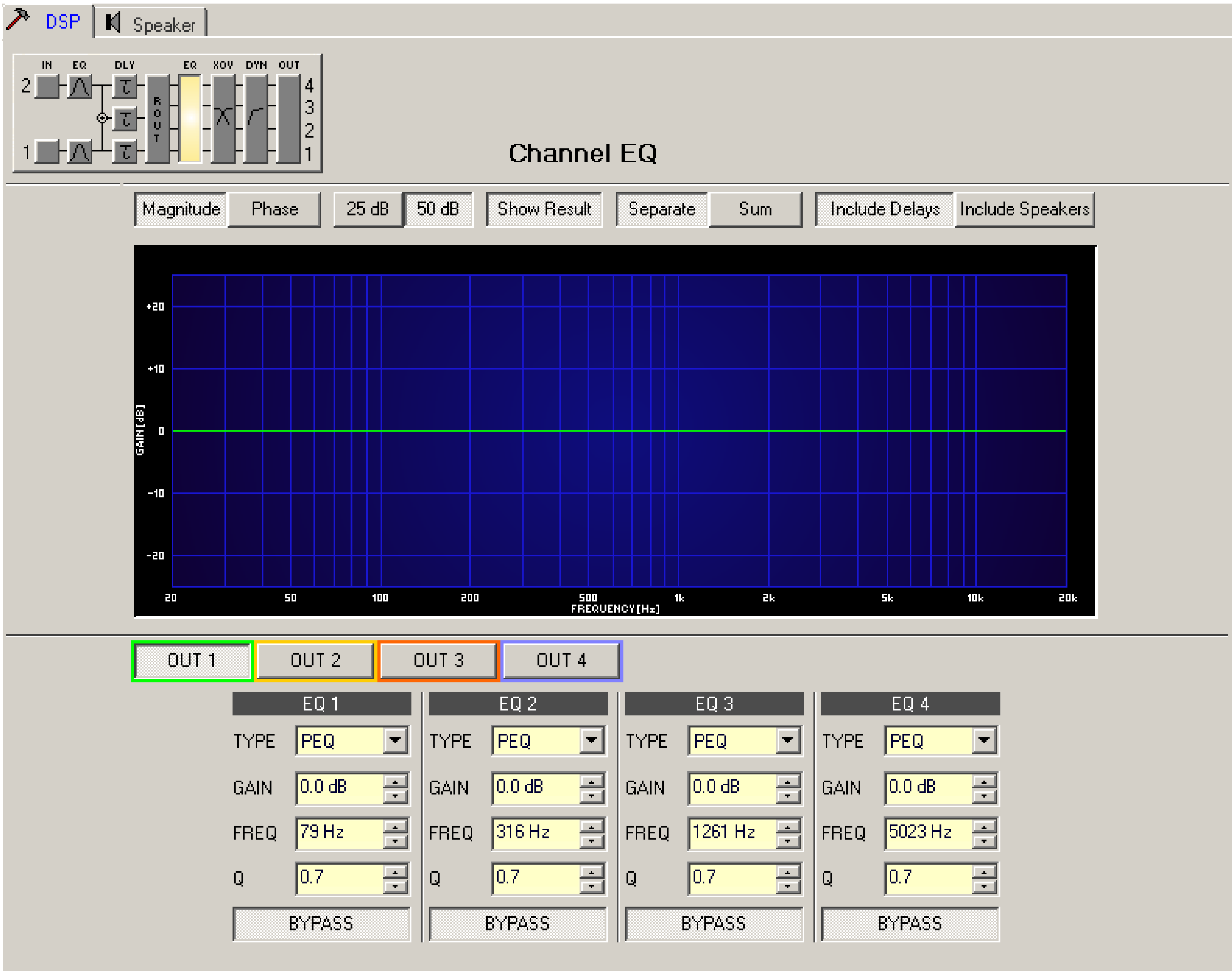
Editing Delays by “Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title “lights” in color as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

CHANNEL EQ

All output channels of the sound system processor employ 4-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select “All pass” as filter type, these filters are identical to the ones of the Master- EQ's.

The Channel-EQ is selected by clicking on the fifth block of the flow diagram selector or by on the CHANNEL EQ block in the full-scale flow diagram.



Graphics Display Indication

Element	Description
Magnitude Phase	Switches between frequency (magnitude) and phase response (phase) indication
25 dB 50 dB	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
Show Result	Shows the resulting transfer function of all filter and level trim settings, the visible and audible result at the sound system processor out- puts. The audible result is displayed in bright colors while “electrical” graphs are indicated in dark colors.
Separate Sum	Selecting ”separate” results in a separated display of the sound system processor channels’ transfer functions while ”sum” shows the summed signal of the sound system processor channels.

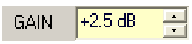
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveal very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the "Speaker" register sheet.

Channel Selection

Element	Description
OUT 1	Switch for selecting output 1, 2, 3 or output 4 for filter editing. A click with the right mouse button opens the "Copy & Paste" menu, which allows convenient copying all EQs of the corresponding output to any other EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the "Copy & Paste" menu, which allows con- venient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE Hipass	PEQ	PEQ. Loshelv. His- helv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmis- sion function.
SLOPE 12dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass fil- ters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
FREQ 80 Hz	79 / 316 / 1261 / 5023Hz	20 Hz...20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q +1.0	0.7	0.4...40.0 (PEQ),	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter,
		0.4...2.0 (Hi-/	while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the

		Lopass), 0.4...2.0 (All- pass)	response of Hi, Lo and All pass filters with slopes of 12dB/oct.
	0 dB	-18...+12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shel- ving and high shelving equalizers.
	first	first, second	ORDER (only available with All pass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via ”Mouse Movement” in the Graphics Display

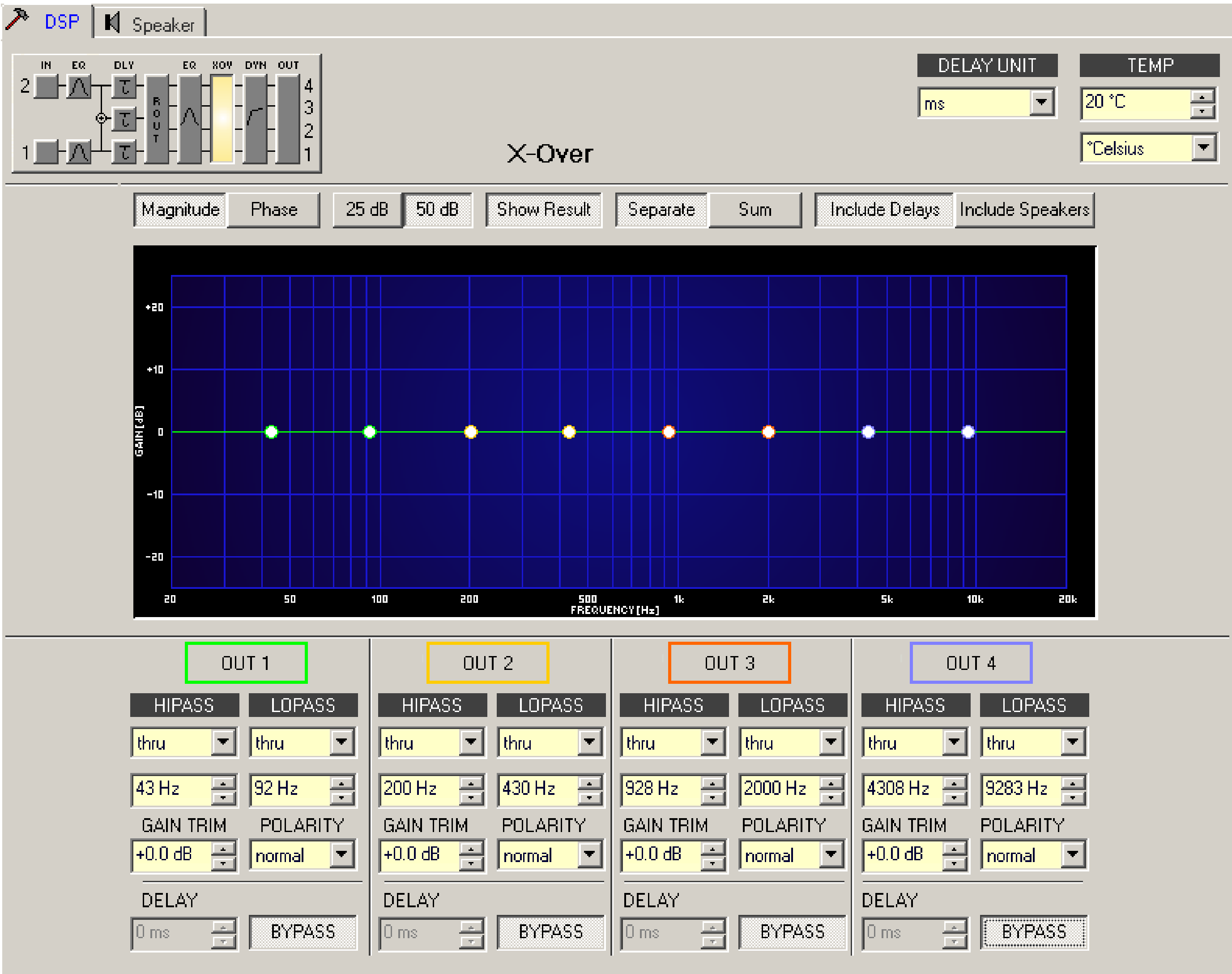
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs.

For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

X-OVER

The X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch, which are provided for each output channel of a sound system processor. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.

Clicking on the sixth block in the Flow Diagram Selector or on the X-OVER block in the large signal flow diagram opens the X-Over window.



Graphics Display Indication

The graphics display offers several different indication modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (Master EQ, Channel EQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Show Result	Displays the resulting transfer function of all filter and level trim settings and therefore the visible respectively audible result at the sound system processor outputs. The audible result is displayed in bright colors while all "electrical" graphs are drawn in dark colors.
Separate Sum	The "Separate" switch allows separate indication of the two amplifier channels' transfer functions. The "Sum" switch causes display of the summed signal of the two channels.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency res- ponse, e.g. as notch filter effect.
Include Speakers	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the "Speaker" register sheet.

Channel Parameters

Element	Default	Range	Description
OUT 1			Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all X-Over parameters of the corresponding output to any other X-Over within the same project.
HIPASS thru 200 Hz	thru, 43 / 200 / 928 / 4308 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes bet- ween 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter fre- quencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI-PASS filters within the same project.
LOPASS thru 430 Hz	thru, 92 / 430 / 2000 / 9283 Hz	RESPONSE: thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB FREQ: 20 Hz...20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes bet- ween 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter fre- quencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button onto the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.

GAIN TRIM +0.0 dB	0 dB	-30 dB...6 dB	GAIN TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual fre- quency bands.
POLARITY normal	normal	normal, inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to”Sum”).
DELAY 0.0 ms	0.0 ms	0.0...900.0 ms	DELAY allows delaying the audio signal of the corresponding output by an adjustable period of time. This delay method is typically used as time-alignment-delay to overcome nega- tive sound effects like they result from different distances between loudspeaker systems within one cabinet or the positioning of speakers in a PA-installation that otherwise would cause a high amount of natural delay.
BYPASS			BYPASS allows activating (button not engaged) or deactivating (button engaged) the corresponding delay.

General Parameters

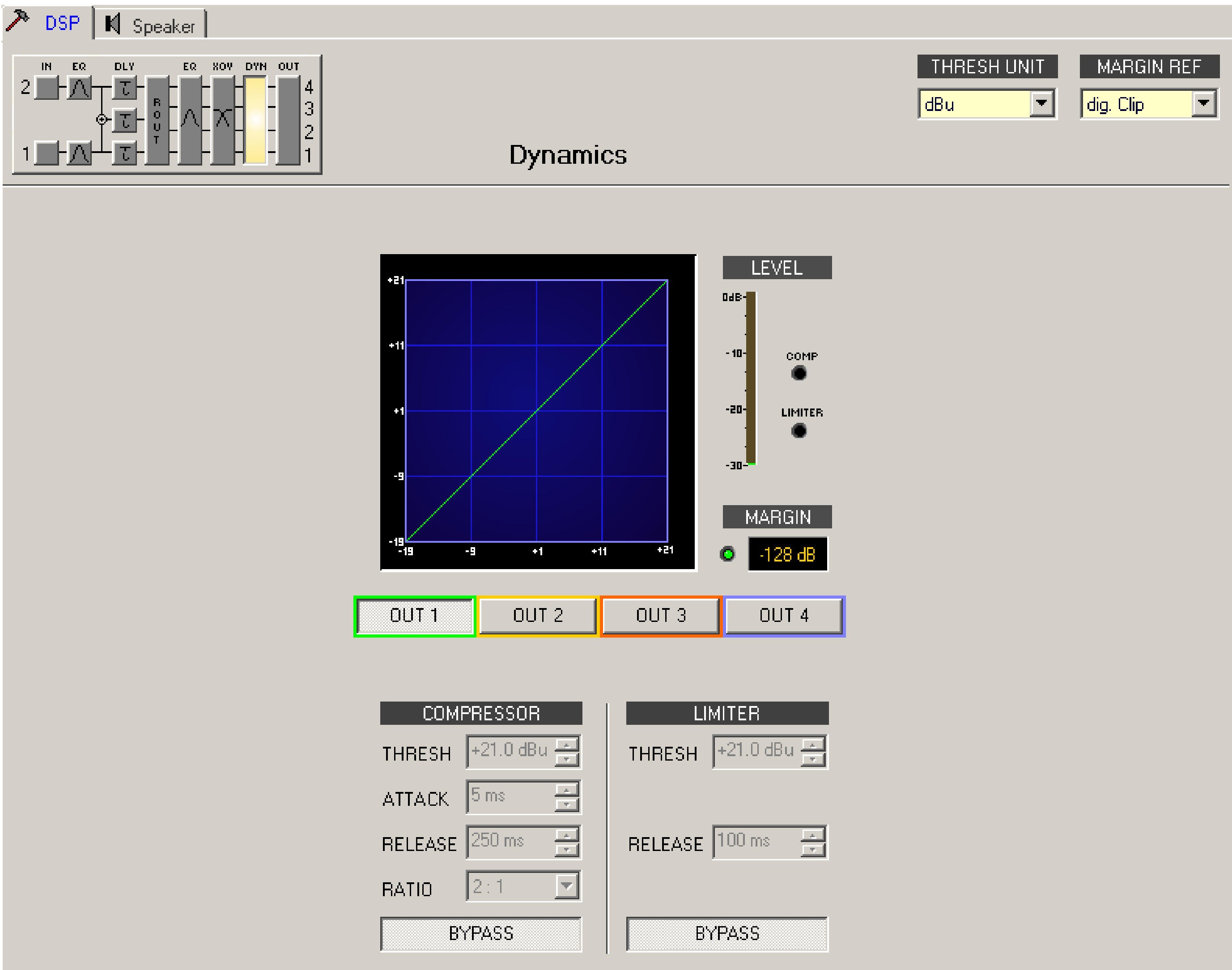
Element	**Default**	**Range**	**Description**
DELAY UNIT m	ms	ms, samples, ft, in, m, cm, µs, s	This lets you select the unit of measurement for the delays.
TEMPERATURE +23 °C °Celsius	20 °C	-20...60 °C -4...140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing X-Over Filters by ”Dragging the Mouse” in the Graphics Display

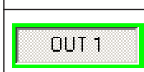
Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right. A filter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide impro- ved overview and handling. An extra white graph is displayed in addition, representing the frequency response of the corresponding selected filter.

DYNAMICS

Each output channel of the sound system processor offers a compressor and a limiter. These functions can be accessed via the dynamics window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.
Clicking onto the seventh block in the Flow Diagram Selector or double clicking onto the DYNAMICS block in the large flow diagram opens the dynamics window.



Channel Parameters

Element	Description
	Channel name A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all dynamics parameters of the corresponding channel to any other channels within the same project.

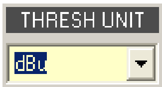
Compressor Parameters

Element	Default	Range	Description
COMPRESSOR			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all compressor parameters of the corresponding channel to any other channels within the same project.
THRESH +55.7 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	The THRESHOLD parameter determines the audio signal level above which the compressor starts operating.
ATTACK 5 ms	5 ms	0...99 ms	ATTACK determines how fast the compressor reduces amplification when the threshold is exceeded.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the compressor returns to normal amplification, after the audio signal level declined the threshold.
RATIO 4 : 1	2:1	1: 1, 1.4: 1, 2: 1, 4: 1, 8:1	RATIO determines the amount of compression that the audio signal is compressed when exceeding the threshold. The setting of 4 : 1 for example relates to a reduction of the audio signal by the factor 4.
BYPASS			BYPASS switches the compressor on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the compressed and non-compressed audio signals.

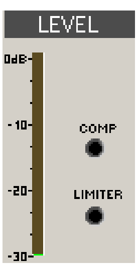
Limiter Parameters

Element	Default	Range	Description
PEAK LIMITER			A click with the right mouse button on this field opens the Copy & Paste menu, which allows copying all limiter parameters of the corresponding channel to any other channels within the same project.
THRESH +55.7 dBu	21 dBu	-9.0...+21.0 dBu or 0.27...8.70 V	THRESHOLD determines the audio signal level above which the limiter starts operating.
RELEASE 250 ms	250 ms	50...999 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
BYPASS			BYPASS switches the limiter on (button is not engaged) or off (button is engaged). This allows quick A / B-comparison of the limited and non-limited audio signals.

General Parameters

Element	Default	Range	Description
	dBu	dBu / Volts	This lets you select the unit for the threshold parameter. The selected setting applies to compressor and limiter as well.
	dig. Clip	dig. Clip, Limiter Thresh	This lets you set the absolute level for margin indication. You can select between ”Digital Clip” (relates to +21 dBu) and ”Limiter Threshold”. The margin level indicates the distance between signal level and the set absolute level. The displayed mar- gin always relates to the highest actual signal level reading.

Indications

Element	Description
	These indicator shows the reduction in dB that is applied to the audio signal by the compressor (COMP) or limiter. Level reduction is indicated as ver- tical yellow bar graph.
	The margin level indicates the distance between signal level and the set absolute level. The displayed margin relates to the highest actual signal level reading since the last reset of the indicator. The LED changes from green to red as soon as the signal level reaches or exceeds the set absolute level (Digital Clip / Limiter Threshold). A click with the right mouse button onto the margin level followed by click onto Reset re-sets indication.

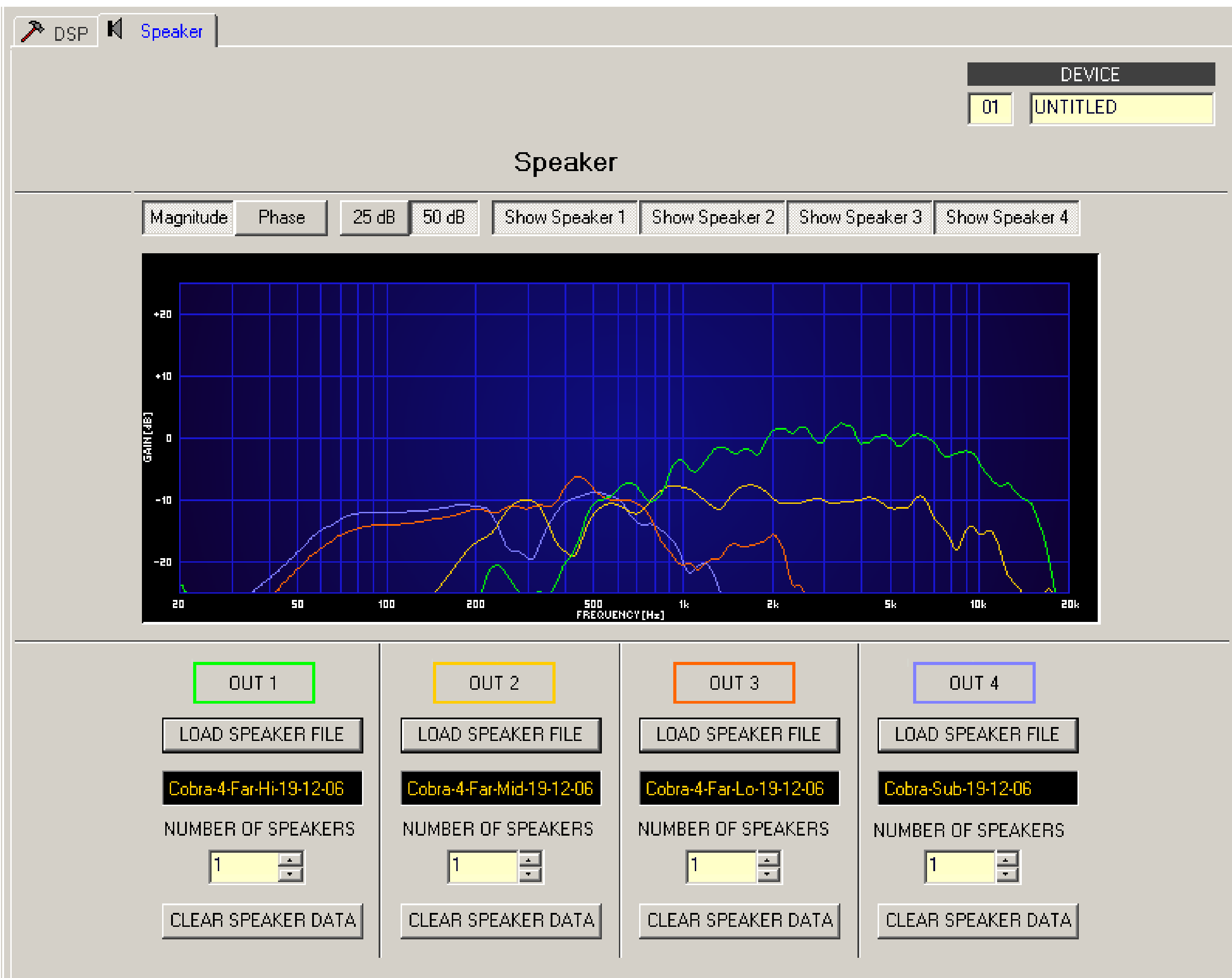
Editing Compressor / Limiter Parameters by ”Dragging the Mouse” in the Graphics Display

Active compressors or limiters (bypass button is not engaged) are indicated by a white dot in the graphics display re- presenting its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding compressor or limiter by vertically dragging the mouse. A click with the right mouse button onto the white dot of a compressor and keeping the mouse button pressed down lets you edit the ratio of compression.

A compressor’s / limiter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

Speaker

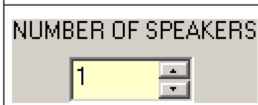
The Speaker Dialog offers the possibility to load the datasets of different loudspeaker systems, assign it to the sound system processor channels and display the acoustic results. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of loudspeaker systems. The speaker data as well as any settings made in this window have no direct influence on the transfer function of the sound system processor. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of sound system processor plus speaker transfer functions.
Clicking on the Speaker tab in the Setup & Control window opens the Speaker page.



Indication on the Graphic Display

Element	Description
	Switch for toggling between frequency response (magnitude) and phase response (phase) display
	Switch for adjusting the scale of the amplifier axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Switching the display of the corresponding speaker data for an sound system processor channel on/off is performed using the ”Show Speaker 1” to ”Show Speaker 4” switches.

Channel Parameters

Element	Default	Range	Description
			Channel name.
			Clicking the button LOAD SPEAKER FILE opens a dialog that allows the selection of the desired speaker file.
			The name of the loaded loudspeaker model is shown in the black-shaded field.
	1	1...8	The NUMBER OF SPEAKERS parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel. Setting an amount from 1 to 8 is possible.
			Clicking the CLEAR SPEAKER DATA button clears the previously loaded measured speaker data of the selected channel.

DSP 600 FIR-TUNE

The DYNACORD DSP 600 Digital System Processor is a universal two-input, six-output digital signal processor with the configuration flexibility to handle a multitude of audio system needs and applications; installed sound, house of worship, convention & meeting facilities, concert touring, club, portable sound reinforcement and more.
The internal signal processing structure can be configured as 2-way stereo + full-range, 3-way stereo, 4-way mono + full range, 5-way mono + full range, 3-way stereo with a mono sub + full-range, 4-way stereo with mono sub and low frequency and finally as a freely assignable 2 x 6 matrix router.
The DSP 600 replaces entire racks of signal processors previously needed to properly configure and control sound reinforcement systems with a single Dual-Core DSP processor. The substantial advantages of the DSP 600 over discrete signal processing racks include:

- 24-bit, 48 kHz digital signal path
- No patch cables to fail or add noise
- Optimal gain structure throughout all stages of signal processing; no gain matching from processor to processor
- Recallable factory and user presets; instant system reconfiguration for differing applications and performances
- Easy, intuitive operation and editing with a PC and IRIS-Net

FIR-TUNE
The DSP 600 includes Finite Impulse Response (FIR) filters at each output for loudspeaker linearization. Using FIR filters has the following advantages, compared to using IIR filters (e.g. Bessel, Butterworth,...).:

- extremely linear frequency response
- very high stop-band attenuation
- linear phase systems

To sum up, FIR-TUNE allows the linearization of frequency and phase of your DYNACORD loudspeakers. Activating FIRTUNE is as easy as loading a FIR Speaker Setting in the output channel of the DSP 600. The IRIS-Net software is used for loading Speaker Settings, and lots of DYNACORD FIR Speaker Settings are included. Please refer to the documentation of IRIS-Net for more details about using Speaker Settings.
Each DSP 600 Digital System Processor includes the following signal processing blocks:

INPUTS

- Pilot tone detection
- VU Metering of input signal
- Analog or digital (AES/EBU) Inputs
- 24-bit, 48 kHz A/D converters
- 10-band parametric equalizer

- 31-band graphic equalizer
- Delay

MATRIX ROUTER / MIXER
- Two inputs (stereo)
- Summed left / right (mono) input
- Six assignable outputs

OUTPUTS (EACH)
- Array control (5-band equalizer +delay)
- Cross-over (hi-pass / low-pass filters), with selectable filter types
- 6-band parametric equalizer
- FIR filter with 512 Taps
- Delay
- Polarity
- Look-ahead Peak limiter with Peak RMS detection
- TEMP Limiter for long-term loudspeaker protection
- Level & Mute
- 24-bit, 48kHz D/A converters
- Pilot tone generator
- VU Metering
- Output assignment display LEDs; sub, low, mid & high
- Mute button
- Gain reduction meters

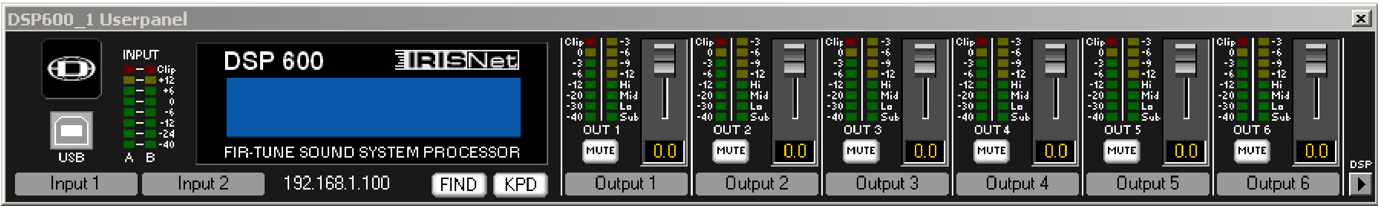
ADDITIONAL FEATURES INCLUDE:
- Electronically balanced XLR inputs and outputs
- XLR thru connectors (analog + AES/EBU)
- -6 dB switchable input level PAD
- Test generator (Sine, pink noise, white noise)
- Contact closure interface
- USB port (front) and Ethernet port (rear) for connection to PC with IRIS-Net software; preset editing and real time parameter control and monitoring.
- Firmware updates via USB port or Ethernet port
- FLASH memory for preset storage and firmware upgrades
- 192 x 32 back-light graphic LCD display
- LCD navigation / editing controls
- DSP block direct access controls
- Auto-ranging internal power supply; 100-240 V AC, 50-60 Hz
- Standard IEC A.C. inlet with external, replaceable fuse

DSP 600 Device

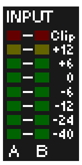
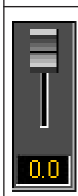
Start by creating an DSP 600 Device in your IRIS-Net project. Drag an DSP 600 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the Ctrl key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an DSP 600 device icon opens the Userpanel.

DSP 600 Userpanel

The DSP 600 Userpanel provides access to the controls and indications on the DSP 600 front panel.



Indications and Functions of the DSP 600 Userpanel

Element	Description
	The level meter displays are meant for visual monitoring of the input signal levels, individually showing the peak value of the correspondent input signal in dBu. The input control should be set to a position so that the meter instruments indicate a level between -6 and -12 dB. To prevent internal clipping, make sure that the CLIP LEDs are not lit.
	In online mode the LCD display is identical on the DSP 600 Userpanel and the device.
	These LEDs indicate the peak level of the corresponding outputs. The level is shown as headroom relative to the D/A clip or limiter threshold, as selected in the DSP 600 menu. The DSP 600 should be operated in a range, so that the clip-LEDs are not lit. Otherwise, this could lead to internal clipping.
	Each output channel has a four-segment gain reduction meter that shows the gain reduction of the output channel Limiter on output signal; from -3dB to -12dB.
	Each output channel has a four-segment function display for informational purposes only. For any given configuration pos- sible with the DSP 600, an output channel may be identified as a sub, low, low/mid, mid, mid/hi, hi or full range output. One or two adjacent LED are displayed to indicate all possible output bandpasses. (Full range is indicated by no lit LED's.)
	Each output channel has a lighted Mute button. Pressing the Mute button turns off the output of that channel. The button lights red as an alert. Press the Mute button again to restore the output channel's signal.
	These controls are used to set the output levels of channels 1 to 6, allowing the matching of the DSP 600 to the input levels of the devices chained in sequence. Correctly setting these controls results in an improved S/N ratio. The digital output gain control should be used when higher output levels are needed. Use the controls to attenuate the output levels. It is not recom- mended to use the digital output gain control for massive attenuation, since this would reduce the dynamic range of the D/A-converters.
	Label of input or output channels, the labels can be edited at the Config & Info window.

	Clicking on the DSP button opens the Configuration Panel, which provides access to all DSP and speaker parameters.
192.168.1.100	Indicates the IP address of the DSP 600's Ethernet port (factory setting: 192.168.1.100). Click to edit the address.
FIND	A click on the FIND button lets the LEDs on the front panel of the DSP 600 blink. When on-line, this allows for easy identifica- tion of which DSP 600 the user is currently communicating with. Click on the FIND button again to stop the LEDs blinking.
KPD	A click on the KPD button opens the Keypad dialog. When on-line, the buttons in the Keypad dialog have the same function as the buttons on the front panel.

Keypad

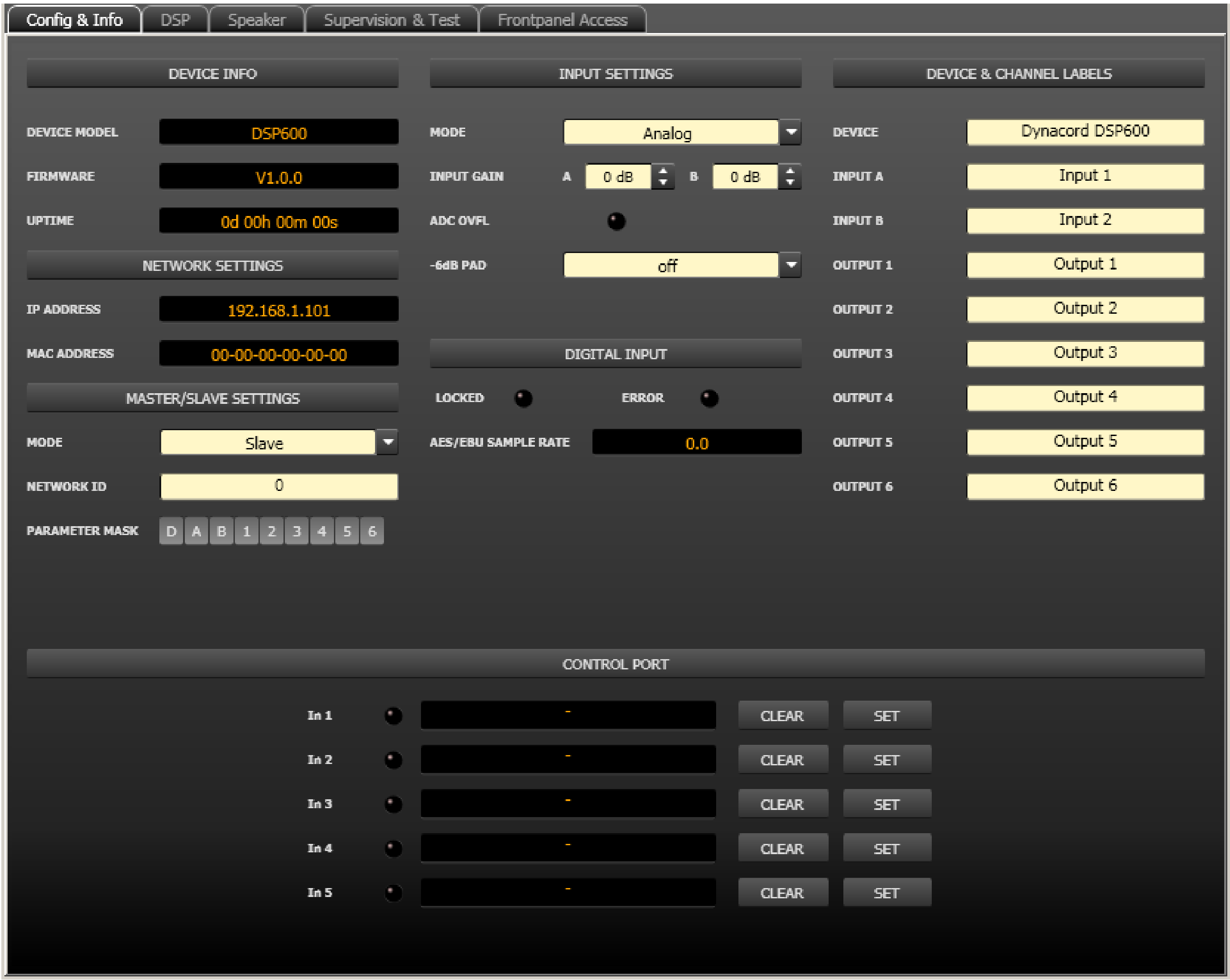
Configuration Panel

Clicking on the „SET" soft key in the DSP 600 Userpanel or selecting the entry DSP600 UI Dialog from the context menu of the device opens the Configuration Panel. The Configuration Panel allows configuration of all DSP 600 parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

Dialog	Description
Config & Info	This page provides information about the DSP 600 and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Input, Array and Speaker) of the DSP 600.
Speaker	This page allows the loading and displaying speaker data.
Supervision & Test	This page provides access to several settings for test generator and pilot tone detection.
Frontpanel Access	This page allows selection of which parameters should be visible or editable from the front panel.

Config & Info

The Config & Info window provides information and basic settings for the selected DSP 600. Additionally, editing labels and configuration of control port functions is possible as well.



Device Info

Element	Description
DEVICE MODEL	Shows the signal processor type
FIRMWARE	Shows the software‘s version number.
UPTIME	Shows the uptime of the DSP 600.

Network Settings

Element	Default	Description
IP ADDRESS	192.168.1.100	IP address of the DSP 600
MAC ADDRESS		MAC address of the DSP 600

Master/Slave Settings

Element	Default	Description
MODE	off	The master/slave settings only work if more than one DSP 600 device is connected to a network. Devices that are Master or Slave have always identical parameter settings. Select „Master“, if this DSP 600 should write the parameter settings to one or more other DSP 600 (Slave). Select „Slave“, if the parameter settings of this DSP 600 should be read from another DSP 600 (master). Select „off“, if the parameter settings of this DSP 600 should be independent from other devices.
NETWORK ID	0	Each master DSP 600 connected to the network must have an unique network id. Enter the id of the Master DSP 600 the parameters should be read from if this DSP 600 is used as „Slave“. Multiple DSP 600s can be Slaves to a single Master, if desired.
	all groups selected	If the MODE „Slave“ is selected, choose the parameter groups that this DSP 600 should read from the Master DSP 600. Following groups are available: D: Parameters of the device A or B: Parameters of input A or B 1 to 6: Parameters of output 1 to 6

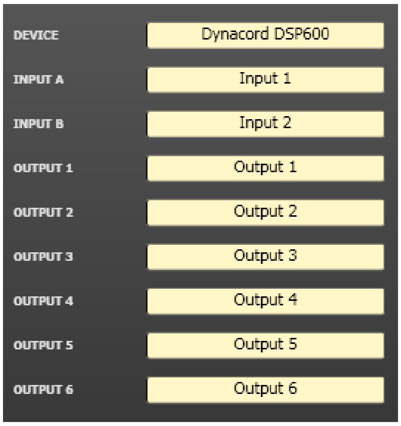
Input Settings

Element	Default	Range	Description
MODE	Analog	Analog, AES/EBU	Select the analog or the digital (AES/EBU) audio inputs of the DSP 600.
INPUT GAIN	0 dB	-60 to +12 dB	Adjust the input gain of the audio input.
ADC OVFL			The LED lights red for 2 seconds if there is an overflow of the A/D converter.
-6dB PAD	off	on, off	Input levels to the DSP 600 can be reduced 6dB prior to the A/D converter to compensate for higher-level output from mixers and other audio devices. For ideal signal to noise performance when connecting the DSP 600 to high output level devices engage the 6dB PAD („on“) rather than turning down the output of the connected device.

Digital Input

Element	Default	Range	Description
LOCKED, ERROR			If the LOCKED LED lights green, the input is synchronized to the incoming signal and the audio is correctly transmitted. When signal transmission fails, the ERROR LED lights red.
AES/EBU SAMPLE RATE	-	32 to 192 kHz	Shows the sampling rate of the incoming signal when the input has been successfully synchronized.

Device & Channel Labels

Element	Description
	The labels of the DSP 600 and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately reflected in the different panels and windows (User panel, flow diagram). The DEVICE label is indicated on the display at the DSP 600 front panel. **CAUTION: Using * (asterisk) and / or = (equal) signs in a name is not permissible.**

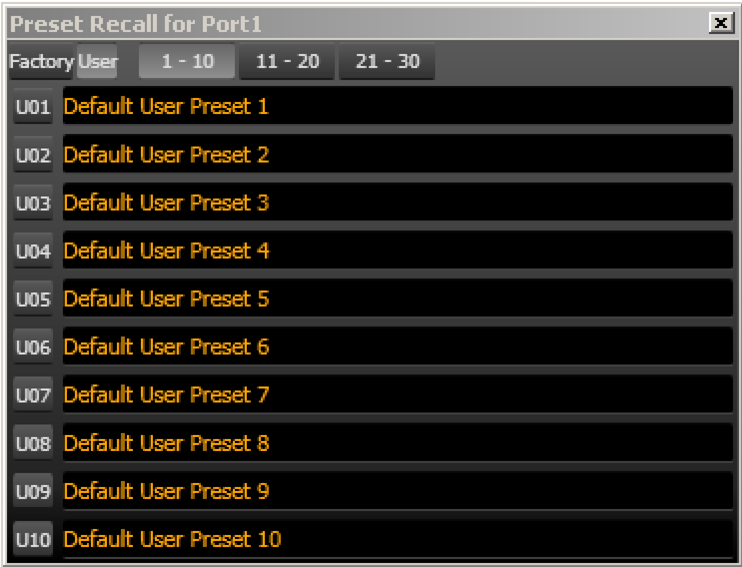
Control Port

The Control Port of the DSP 600 provides five control inputs and a reference connection for ground. The control inputs can be used for the recall of presets. For more information and electrical specifications of the control port, please refer to the DSP 600 manuals.

Element	Description
	Description and current state of the input. The LED lights green if the input is connected to ground.
	Description of the preset to be recalled by the input.
	Clears the preset assignment to the input.
	Opens the Preset Recall for Port x dialog. This dialog allows assignment of a factory or user preset to the input.

Preset Recall for Port x

This dialog lists the 60 Factory Presets and 30 User Presets of the DSP 600.



Element	Description
Factory / User	Switch between the Factory Presets or User Presets.
1-10, 11-20, 21-30, ...	Select the preset group that should be listed.
U01...U10	Press the button of the preset that should be assigned to the input.

DSP

The DSP pages provide overview and access to all DSP parameters of the sound system processor. Within this window you can use the Flow Diagram Selector to link to different function groups.

Element	Description
 	The current preset is in Edit mode if the yellow EDIT button is indicated. Pressing the EDIT button „compares“ the edited preset, if parameters have been altered, to the original unedited preset. This compare function will audibly switch between the altered parameters and the previously stored settings, allowing you to hear the effect of any DSP changes that have been made. Use this feature to monitor progress in editing or creating presets. Subsequently recalling a new preset will prompt you to save changes, which you may do or not.

FLOW DIAGRAM SELECTOR

The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a light grey engaged field.

A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

Element	Description
Flow Diagram	The signal flow display provides an overview of the DSP settings. This area also includes all controls for the preset location/file management and configuration settings.
Input Parametric EQ	The Input Parametric EQ page provides access to the two 10-band parametric equalizers of the sound system processor inputs.
Input Graphic EQ	The Input Graphic EQ page provides access to the two 31-band graphic equalizers of the sound system processor inputs.
Input Delay	This page allows the programming of delay lines for the input channels A and B.
Array Parametric EQ	The Array Parametric EQ page offers access to the 5-band parametric equalizers of the sound system processor outputs.
Array Delay	This page allows the programming of delay lines for the output channels.
Output Parametric EQ	The Output Parametric EQ page offers access to the 6-band parametric equalizers of the sound system processor outputs.
Output X-Over	Frequency crossover-filters as well as the parameters gain and polarity for all output channels are located in the Output X-Over area.
Output FIR	This page provides a FIR-Filter for each output channel.
Output Delay	This page allows the programming of delay lines for the output channels.
Output Limiters	This page provides access to Peak limiter and TEMP limiter of each output channel.

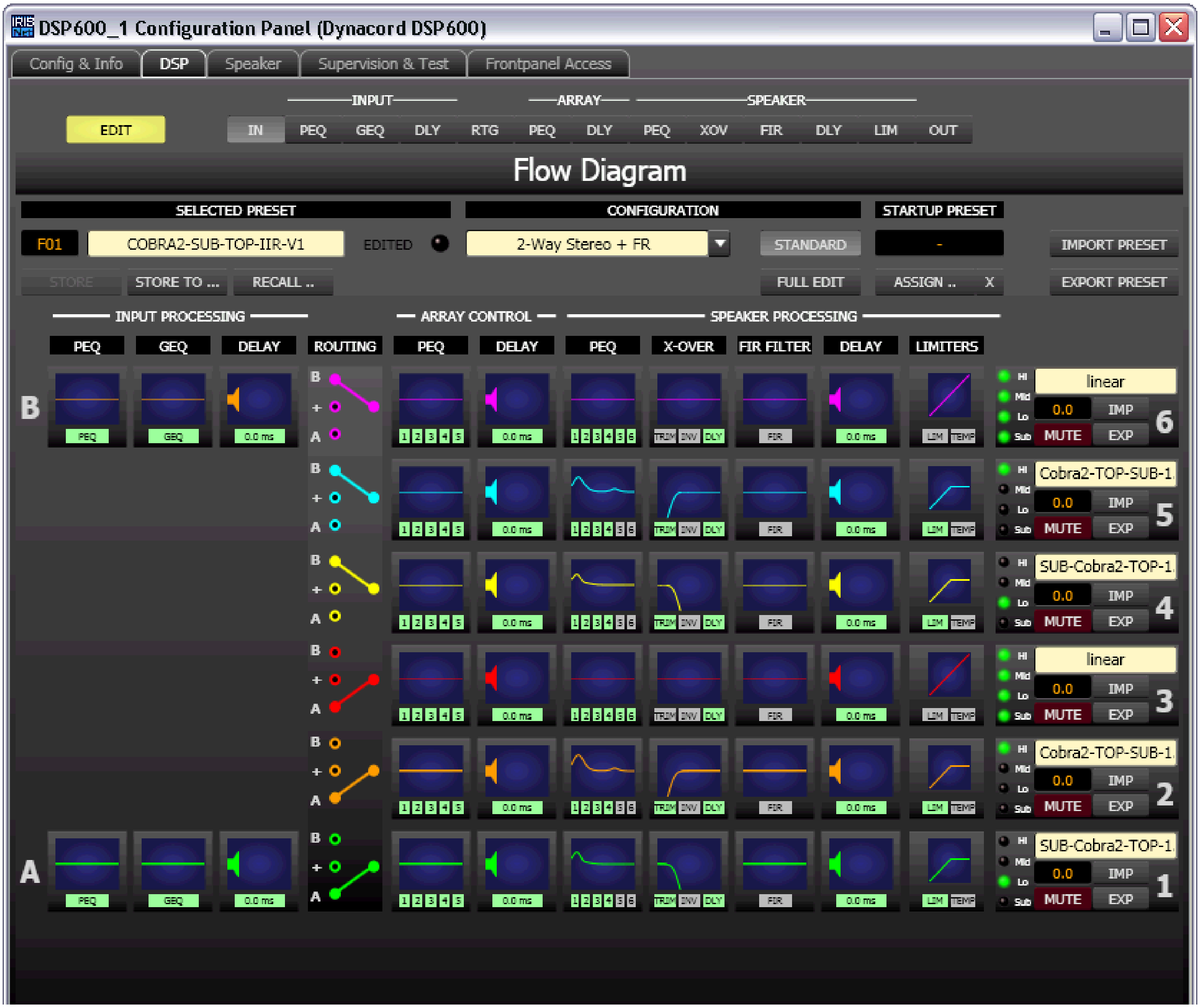
FLOW DIAGRAM

The Flow Diagram window shows a signal flow diagram, which offers a quick overview of all DSP setting of the DSP 600.

- Output muting,
- routing channels,
- setting output level,
- editing configuration LEDs (Free Configuration mode only),
- import and export of Speaker Settings

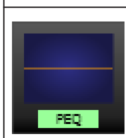
can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for the saving, loading and previewing of presets are also accessible from this window.

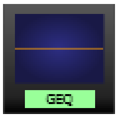
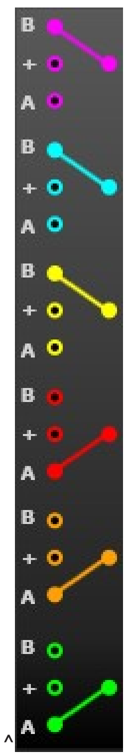
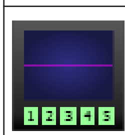
The FLOW DIAGRAM window opens when clicking on the first (IN), fifth (RTG) or 13. block (OUT) in the Flow Diagram Selector.

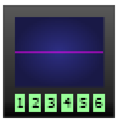
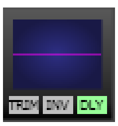
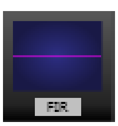
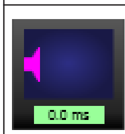


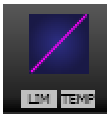
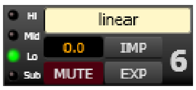
Function blocks

The caption below the function blocks is shown in green if the function, or at least one filter of the block, is activated.

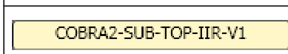
Element	Description
	INPUT PEQ Block: The INPUT PEQ block displays the 10 EQs of the corresponding input channel. The graph shows the frequency response of the EQ block. A single click with the left mouse button onto this block opens the Input Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other DSP 600 INPUT PEQ block within the same project.

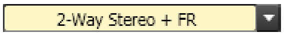
	INPUT GEQ Block: The INPUT GEQ block displays the 31 graphical EQs of the corresponding input channel. The graph shows the frequency response of the GEQ block. A single click with the left mouse button onto this block opens the Input Graphic EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding GEQ block to any other DSP 600 GEQ block within the same project.
	INPUT DELAY Block: This displays the Delay of the input channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Input Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other Input Delay block within the same project.
	ROUTING Block: Here you can assign the output channel routing. The circles next to A and B allow selecting the input signal for the corresponding output channel. The circle next to the + allows selecting the summed input signal for the corresponding output channel.
	ARRAY PEQ Block: The ARRAY PEQ block displays the 5 Array EQs of the corresponding output channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Array Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding ARRAY PEQ block to any other EQ block within the same project.

	ARRAY DELAY Block: This displays the Array Delay of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Array Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other ARRAY DELAY block within the same project.
	SPEAKER PROCESSING PEQ Block: The SPEAKER PROCESSING PEQ block displays the 6 Channel EQs of the corresponding output channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Output Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Speaker EQ block to any other EQ block within the same project.
	SPEAKER PROCESSING X-OVER Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim (TRIM), polarity (INV) and delay (DLY). A single click with the left mouse button onto this block opens the Output X-Over page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other DSP 600 X-Over block within the same project.
	SPEAKER PROCESSING FIR FILTER Block: This block represents the FIR Filter within the corresponding output channel. The graph shows the frequency response that results from the set FIR parameters. The LED indicate if the FIR Filter is being used. A single click with the left mouse button onto this block opens the Output FIR page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding FIR Filter block to any other FIR Filter block within the same project.
	SPEAKER PROCESSING DELAY Block: This displays the SPEAKER PROCESSING DELAY of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Speaker Processing Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other DSP 600 Speaker Delay block within the same project.

	SPEAKER PROCESSING LIMITERS Block: This block provides graphical display of the limiter functions of the corresponding output. The two LEDs indicate whether peak limiter or TEMP limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Limiters block to any other DSP 600 Limiters block within the same project.
	Output Block: For any given configuration possible with the DSP 600, an output channel may be identified as a sub, low, low/mid, mid, mid/hi, hi or full range output. One or two adjacent LED are displayed to indicate all possible output band passes. (Full range is indicated by no lit LED’s.) When Free Configuration is selected the LEDs can be set manually. In online mode the LEDs here and at the front panel are identical. A click with the right mouse button onto OUT 1 to OUT 6 opens the Copy & Paste menu, which allows copying of all parameters of the corresponding output channel to any other DSP 600 output channel within the same project. The numerical field is identical to the numerical field below the level controls in the User panel, click to edit the value. The MUTE button is for attenuating the output level of the corresponding output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute- function and the output is again active. The MUTE button is virtually disengaged and not lit. The IMP or EXP buttons allow import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for the SPEAKER PROCESSING blocks. The text field allows editing the description of the Speaker Setting file to be exported. Importing a Speaker Setting automatically imports the corresponding Speaker File.

Status Indication

Element	**Description**
	Displays the number according to the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	Indicates the name of the actually audible preset. Click to edit the preset name.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

	The DSP 600 offers 7 pre-defined configurations. A configuration is a basic setting that includes the routing of inputs and outputs, the function of the outputs (Sub, Lo, Mid, Hi, Fullrange), as well as basic X-Over parameters. If you do not want to use one of the pre-defined configurations, you have also the possibility to operate the DSP 600 in Free Configuration mode. This mode offers access to all parameters and basically any input/ output routing can be programmed. Additionally, the output assignment (output function - Sub, Lo, Mid, Hi, Fullrange) is also freely definable.
	The DSP 600 defaults to Standard Edit mode wherein input and output channel parameters are appropriately linked. In Full Edit mode, no parameter links are enforced, regardless of the configuration selected. When switching from Full Edit mode to Standard Edit mode, parameters are linked again. At the same time parameters settings in linked channels (upper numbers) will be over- written.

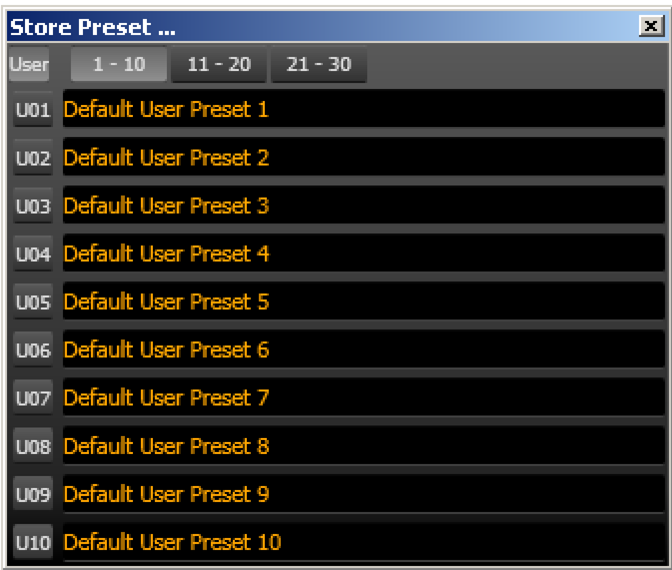
Recall a Preset

Element	Description
	Clicking the RECALL... button opens the Recall Preset dialog, where you can select and recall a Preset.

Caution!
The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

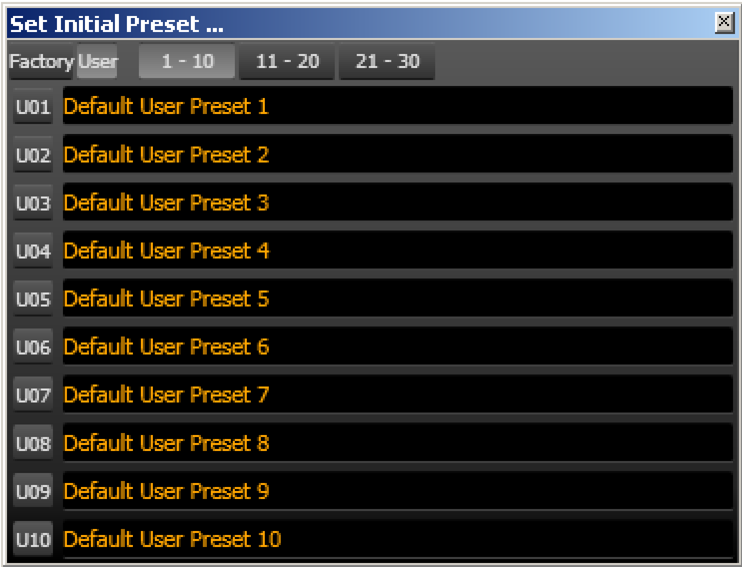
Store a Preset

Element	Description
STORE	STORE saves all currently set DSP parameters together with the entered name into the current preset. Store is only possible if a user program number is selected.
STORE TO ...	Clicking the STORE TO.... button opens the Store Preset... dialog. In this dialog the Program Number can be selected.



Setting a Startup presets

Element	Description
STARTUP PRESET U01	The indicated preset is loaded after power on or reset of the DSP 600.
ASSIGN ..	Clicking the ASSIGN .. button opens the Set Initial Preset... dialog. In this dialog a factory preset or user preset can be selected as startup preset.
X	Clicking the X buttons clears the Startup preset selection.



Import / Export a Preset

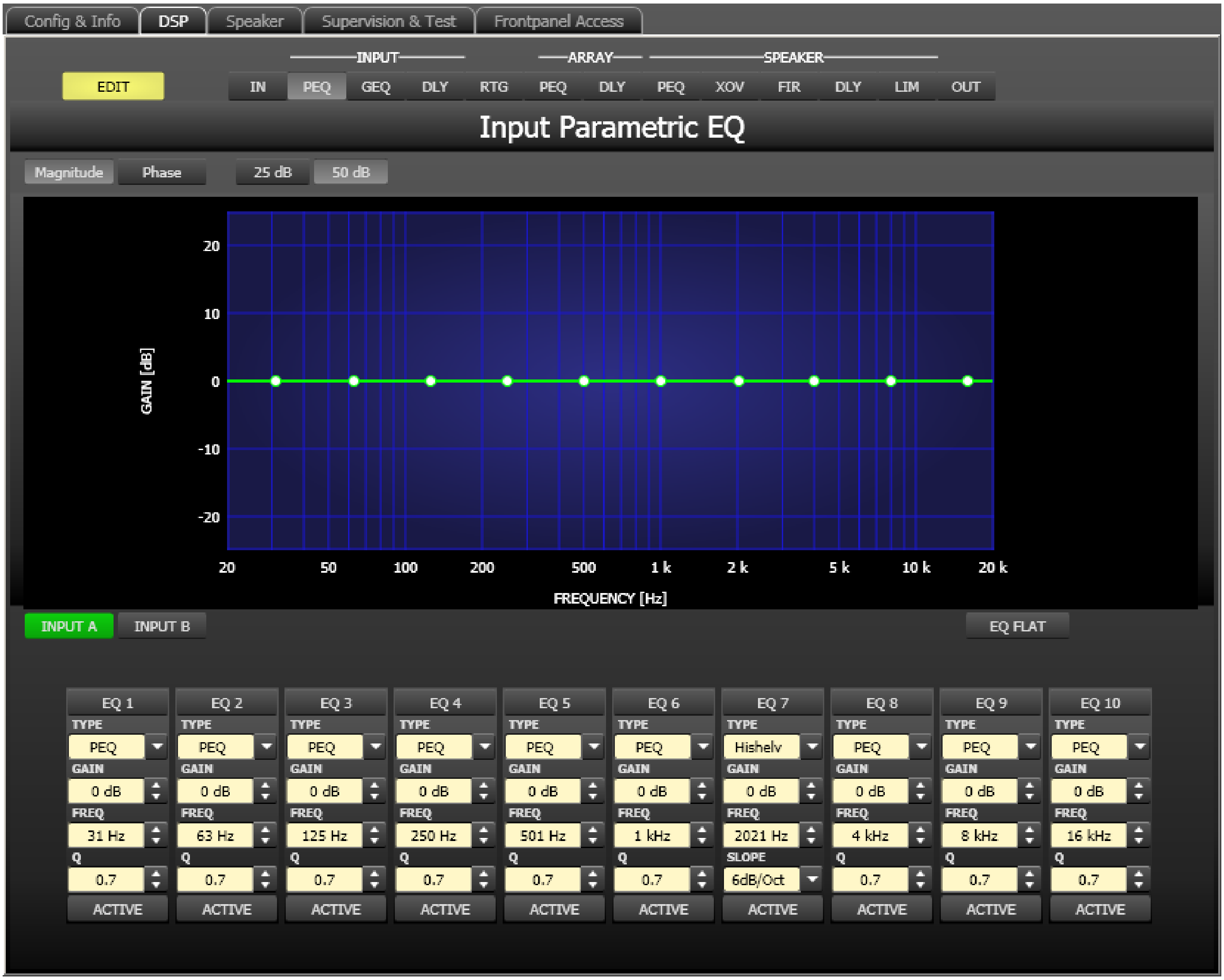
IRIS-Net allows the storing of all DSP parameters of an sound system processor together with the according preset name in a file on a PC, and to load sound system processor parameters from these files. Therefore, IRIS-Net creates a sub- directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory \Presets is permissible.

Element	**Description**
IMPORT PRESET	After clicking onto IMPORT PRESET an ”Open File ...” dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a ”Save File ...” dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click the SAVE button to store all DSP parameters together with the corresponding file name. ”.ds” is automatically added as file extension.

INPUT PARAMETRIC EQ

Both input channels of the sound system processor employ 10-band parametric equalizers each, which allow programming highly variable speaker equalization to match a PA-system to different environmental and acoustical requirements.

The Master-EQ is selected by clicking on the second block (PEQ) of the flow diagram selector or on the INPUT PROCES-SING PEQ block in the full-scale flow diagram.



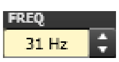
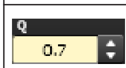
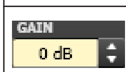
Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

Element	Description
	Switch for selecting input A or input B for filter editing and display. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other DSP 600 Input PEQ filter bank within the same project.

Filter Parameters

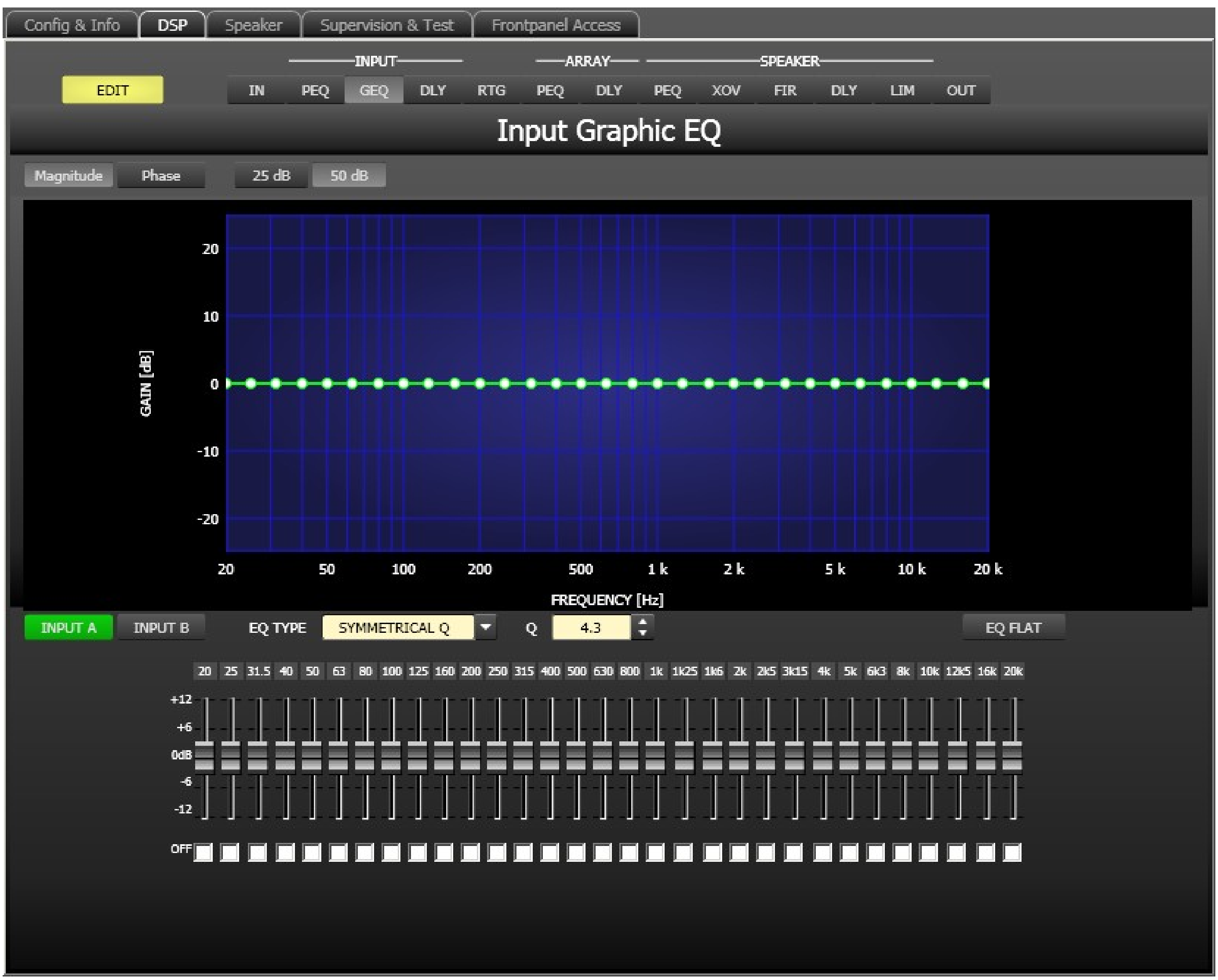
Element	Default	Range	Description
			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
	31 / 63 / 125 / 250 / 500 / 1k / 2k / 4k / 8k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/ Lopass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its magnitude (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs and Hipass/Lopass filters.
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

INPUT GRAPHIC EQ

Both input channels employ 31-band graphic equalizers each, which allow programming highly variable speaker equalization to match a PA-system to different environmental and acoustical requirements.
The Master-EQ is selected by clicking on the third block (GEQ) of the flow diagram selector or on the INPUT PROCES-SING GEQ block in the full-scale flow diagram.



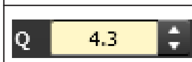
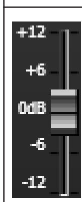
Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

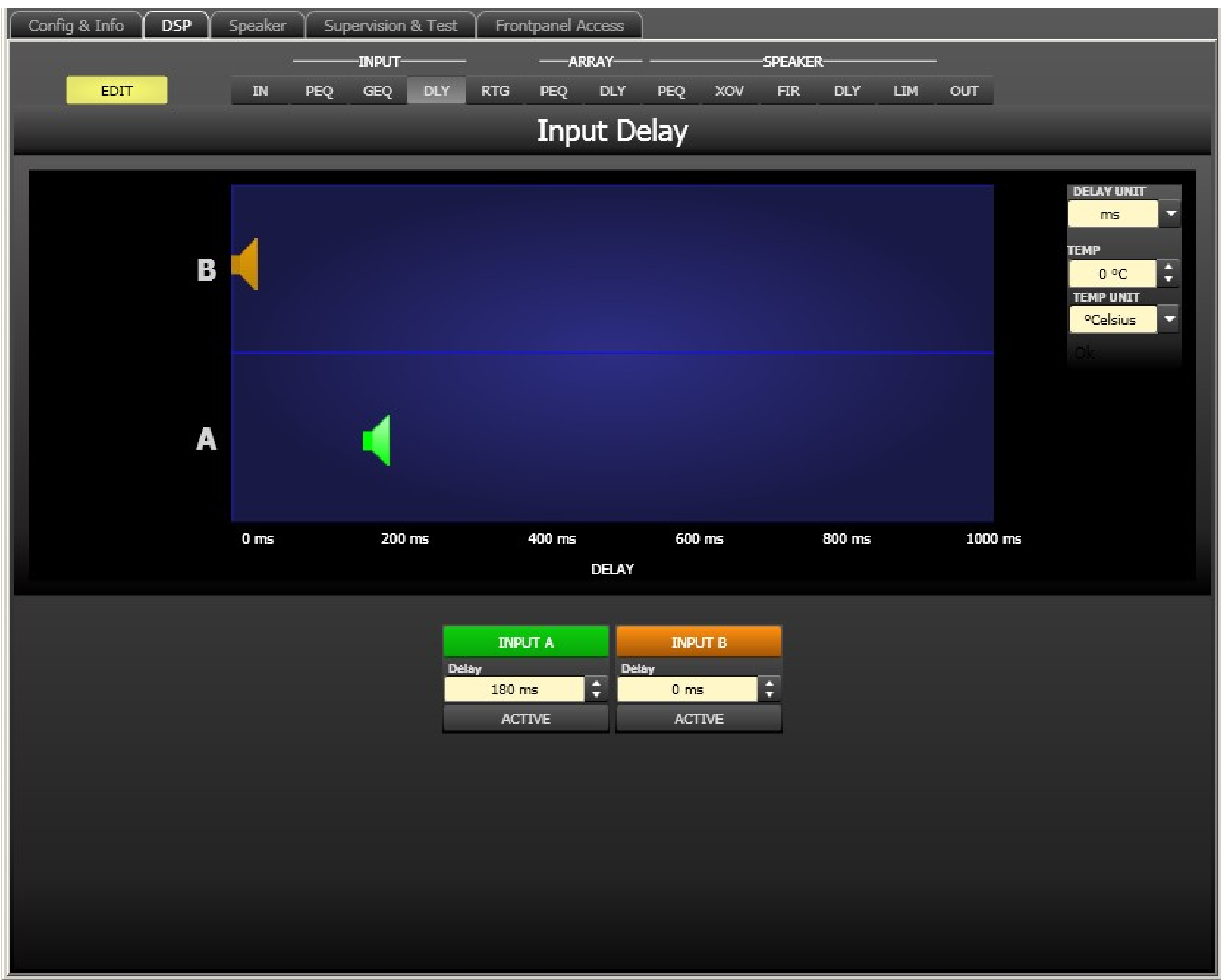
Element	Description
	Switch for selecting input A or input B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other DSP 600 Graphic EQ filter bank within the same project.

Filter Parameters

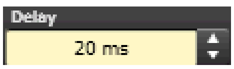
Element	Default	Range	Description
	SYMMETRICAL Q	SYMMETRICAL Q, PROPORTIONAL Q, CONSTANT Q	Switching the graphic equalizer’s type between SYMMETRICAL Q, PROPORTIONAL Q and CONSTANT Q. SYMMETRICAL Q: The filters have an identical Q at all accentuation settings. The lowering frequency responses are symmetrical to the accentuation frequency responses. PROPORTIONAL Q: A Filter’s Q increases as soon as accentuation or lowering of the filter increases, with the effect that the equalizer becomes ”sharper” with increased EQ setting. The quality defined by Q corresponds to the quality at full accentuation or lowering. CONSTANT Q: The filter has the same Q at all accentuation or lowering settings. The resulting accentuation or lowering frequency response is not symmetrical.
	4.3	3.0 to 10.0	Q sets the quality of all EQ bands. A high Q value results in a narrowband filter. A low Q value results in a wideband filter.
			The fixed frequencies of EQ bands
			Sets level amplification (accentuation) or reduction (lowering) of a band. A band’s fader is indicated in red when the band has been deactivated by marking the checkbox OFF. Pres- sing the spacebar resets the currently selected fader to 0 dB.
			Deactivating every single EQ-Band is possible by setting this checkbox. Deactivating a band does not change the band’s previously made settings.
			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via “Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (Checkbox OFF not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s amplification by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-value of the filters by moving the mouse up or down.
For an improved overview the fader of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

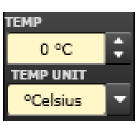
INPUT DELAY
Individual input delays can be set for each input channel of the DSP 600.
HINT: The Input Delay parameter is especially useful for delay lines. In this case the required delay depends only on the position of the delay line and is identical for all ways, e.g. output channels of the DSP 600. By editing the Input Delay parameter the delays of all output channels routed to this input are adjusted automatically.
You can select the input delay window by clicking onto the fourth block (DLY) in the Flow Diagram Selector or onto the INPUT PROCESSING DELAY block in the flow diagram.



Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding input to any other delay within the same project.
	0 ms	0 to 1000 ms	DELAY allows delaying the corresponding input channel’s audio signal by an adjustable period of time. Entering a value only or a value and unit is possible.
			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the input delay.

General Parameters

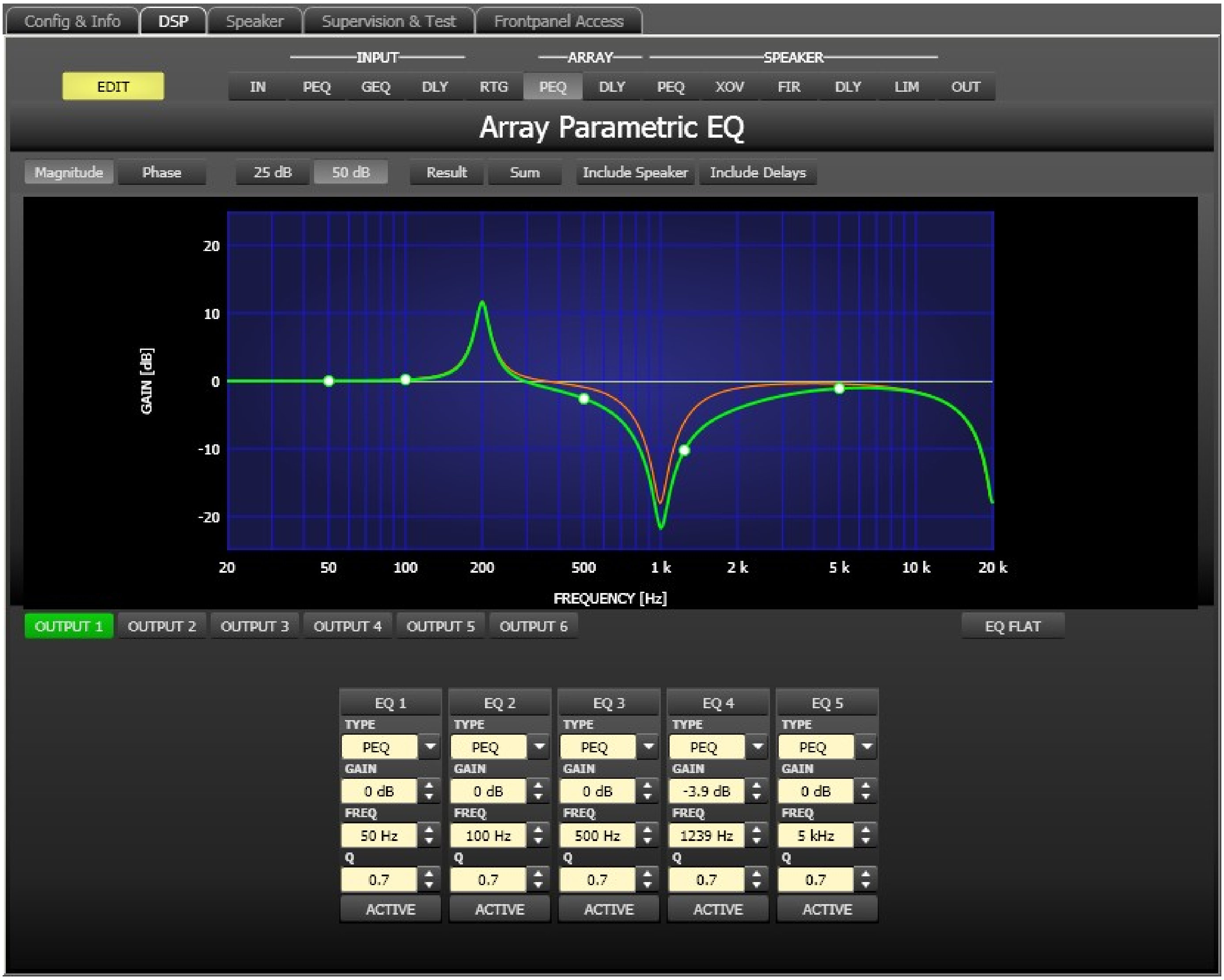
Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

ARRAY PARAMETRIC EQ

All output channels employ 5-band parametric equalizers each, mainly for speaker equalization of arrays. Except for the possibility to select ”All pass” as filter type, these filters are identical to the ones of the input EQ's.
The Array Parametric EQ is selected by clicking on the sixth block (PEQ) of the flow diagram selector or onto the ARRAY CONTROL PEQ block in the full-scale flow diagram.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Array Parametric EQ (input PEQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
	Switch for displaying frequency response (magnitude) or phase response (phase)
	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Displays the resulting transfer function of all filter and level settings and therefore graphically displaying the audible result at the sound system processor outputs.
	The ”Sum” switch causes display of the summed signal of the output channels, including Output level and Mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally displaying measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other DSP 600 Array EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	50 / 100 / 500 / 1k / 5k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..

	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with Allpass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via ”Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

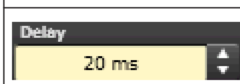
ARRAY DELAY

Individual array delays can be set for each output channel of the DSP 600.

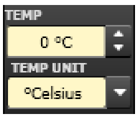
HINT: The Array Delay parameter can be used for adjusting the individual cabinets within a loudspeaker cluster, such as a subwoofer array or a center loudspeaker cluster. For example, in a speaker cluster consisting of two horn-loaded loudspeakers, it is helpful to apply 3-5 ms of delay to one of the loudspeakers in the cluster to improve the coverage in the overlap of the horn patterns. Additionally, the array delay provides a convenient section to apply dedicated delay to individual subwoofer cabinets to create gradient or beam-formed arrays.

You can select the Array Delay window by clicking onto the seventh block (DLY) in the Flow Diagram Selector or onto the ARRAY CONTROL DELAY block in the flow diagram.

Channel Parameters

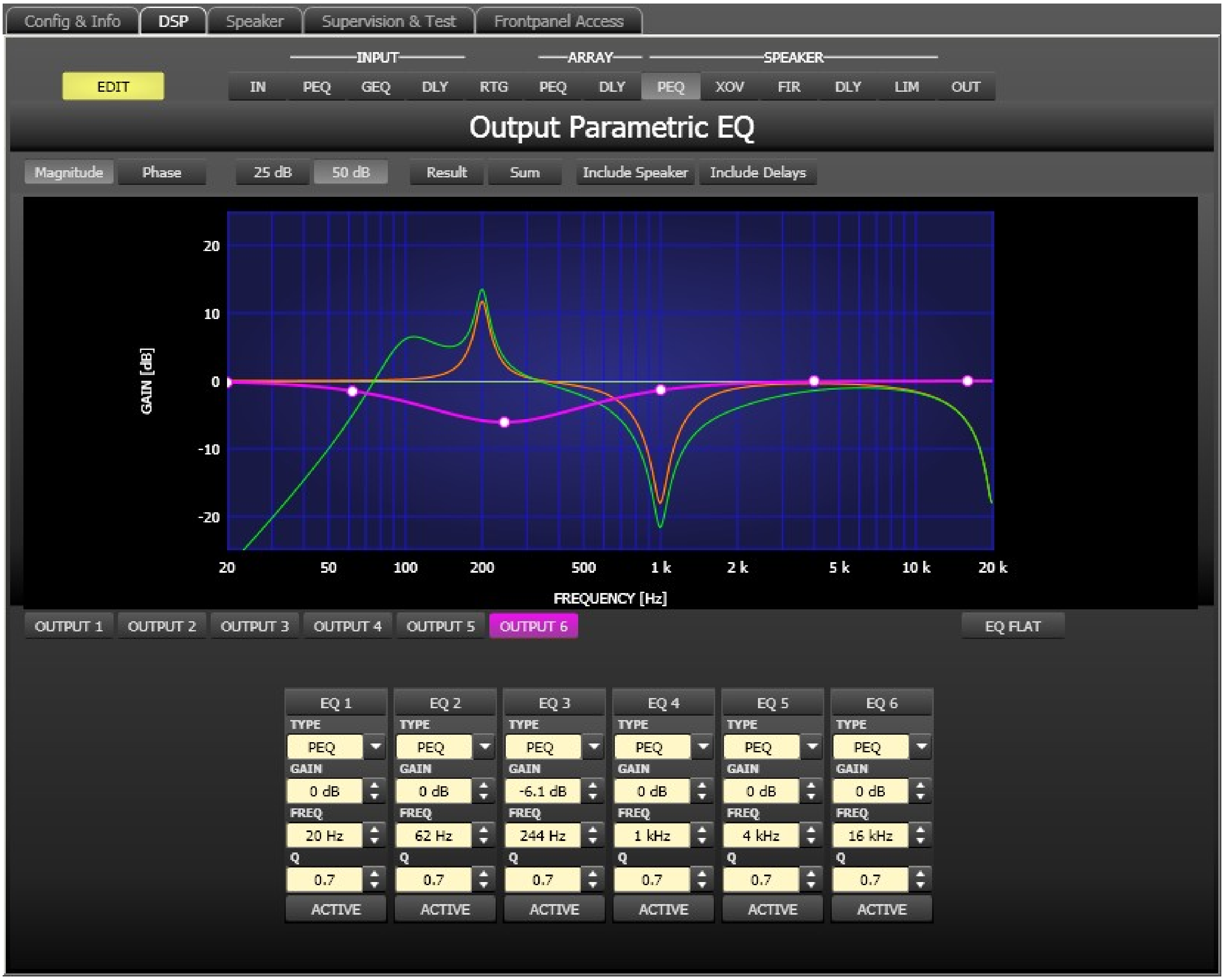
Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
	0 ms	0 to 1000 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the delay.

General Parameters

Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display
The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT PARAMETRIC EQ
All output channels employ 6-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select ”All pass” as filter type, these filters are identical to the ones of the input EQ's.
The Output Parametric EQ is selected by clicking on the eight block (PEQ) of the flow diagram selector or on the SPEAKER PROCESSING PEQ block in the full-scale flow diagram.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Output Parametric EQ, which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
	Switch for displaying frequency response (magnitude) or phase response (phase)
	Switch for scaling the dB- axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs.
	The ”Sum” switch causes display of the summed signal of the output channels, including output level and mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency res- ponse, e.g. as notch filter effect.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other DSP 600 Output EQ-filter bank within the same project.

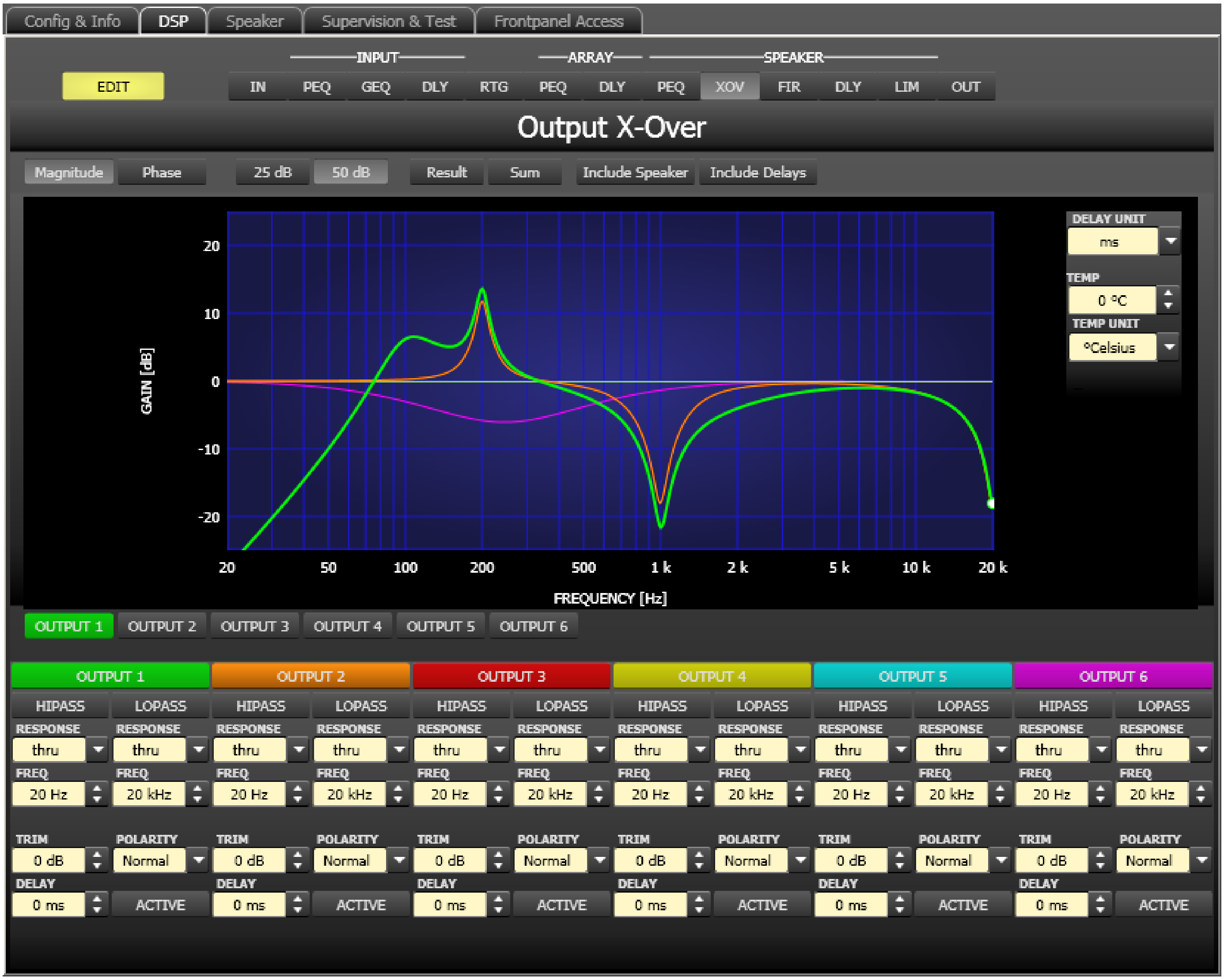
Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	20 / 62 / 250 / 1k / 4k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
GAIN 0 dB	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.

	first	first, second	ORDER (only available with Allpass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the fil- ter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via ”Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

OUTPUT X-OVER
The Output X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.
Clicking on the ninth block (XOV)in the Flow Diagram Selector or on the SPEAKER PROCESSING X-OVER block in the large signal flow diagram opens the X-Over window.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (e.g. Array Parametric EQ), which always provides precise overview and control of the resulting frequency response at this point.

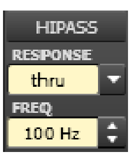
Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.
Sum	The ”Sum” switch causes display of the summed signal of the output channels. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

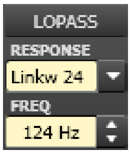
	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency res- ponse, e.g. as notch filter effect.
	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all X-Over settings of the corresponding output to any other X-Over within the same project.

Channel parameter

Element	Default	Range	Description
	thru, 35 Hz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the HI-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI- PASS filters within the same project.

	thru, 16 kHz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the LO-PASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.
	0 dB	-30 dB to 6 dB	TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
	Normal	Normal, Inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to “Sum”).
	0 ms	0.0 to 20 ms	DELAY allows the delay of the audio signal from the corresponding output by an adjustable period of time. **HINT: The X-Over Delay parameter is used for the alignment of transducers within cabinets. Optimized delay values are included in DYNACORD Speaker Settings and should not be edited.**
			Press the ACTIVE button to deactivate the delay (Bypass), which allows for quick A / B-evaluation of the actual effect that the delay has on the sound.

Editing X-Over Filters by ”Dragging the Mouse” in the Graphics Display

Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right. A filter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

OUTPUT FIR

Each output of the DSP 600 offers a 512 taps FIR filter.

The Output FIR is selected by clicking on the tenth block (FIR) of the flow diagram selector or on the SPEAKER PROCESSING FIR block in the full-scale flow diagram.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.
Sum	The ”Sum” switch causes display of the summed signal of the output channels. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency res- ponse, e.g. as notch filter effect.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

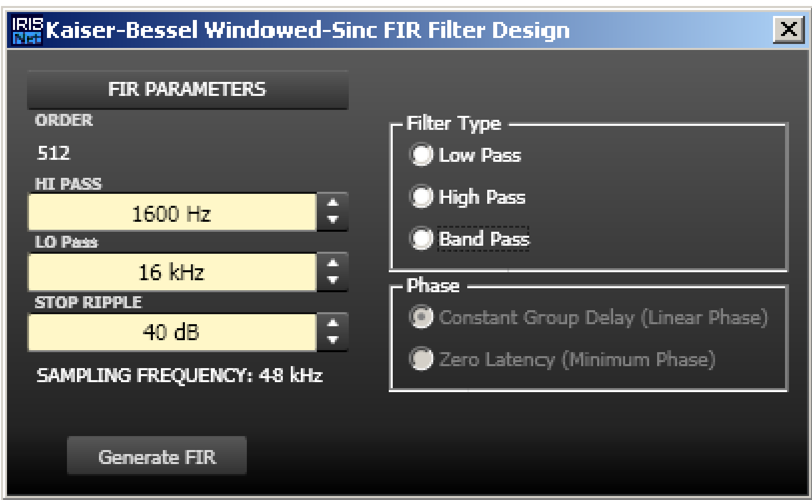
Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying of all FIR filter settings of the corresponding output to any other FIR filter within the same project.

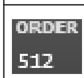
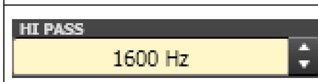
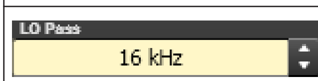
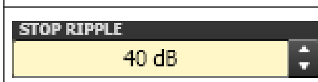
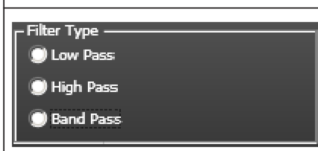
Channel Parameters

Element	Description
FIR INFO IRIS-Net FIR-Filter=Thru	Description of the FIR filter currently in use.
LOAD	After clicking onto LOAD the "Open File..." dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT	After clicking on EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click on the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
CLEAR	Clears the current FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
NEW	Clicking on the NEW button opens the Filter Design dialog.
ACTIVE	Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that the filter has on the sound.
TRIM	Adjusts the gain of the signal between -30 dB and +6 dB.

	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
	Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the output is again active. The MUTE button is virtually disengaged and not lit.

FIR-Filter Design



Element	**Default**	**Range**	**Description**
			ORDER of the FIR filter.
	1600 Hz	20 to 19999 Hz	HI PASS sets the cut-off frequency of the Hi pass filter.
	16 kHz	21 to 20000 Hz	LO Pass sets the cut-off frequency of the Lo pass filter.
	40 dB	21 to 100 dB	STOP RIPPLE sets the slope of the FIR filter.
			Allows selection of the FIR filter type of the corresponding output channel.
			Press this button to generate the FIR filter.

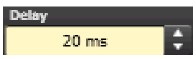
OUTPUT DELAY

Individual output delays can be set for each output channel of the DSP 600.

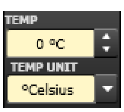
HINT: The DSP 600's output delays can be used to compensate for the positioning of cabinets or speaker arrays relative to each other or the original sound source, for example aligning the PA to the stage or aligning the full-range loudspeakers to the subwoofers. The Output Delay parameter determines the delay time of the corresponding channel or the distance between different loudspeaker clusters.

You can select the output delay window by clicking onto the eleventh block (DLY) in the Flow Diagram Selector or on the SPEAKER PROCESSING DELAY block in the flow diagram.

Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
	0 ms	0 to 1000 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
			Press the ACTIVE button to deactivate the output delay.

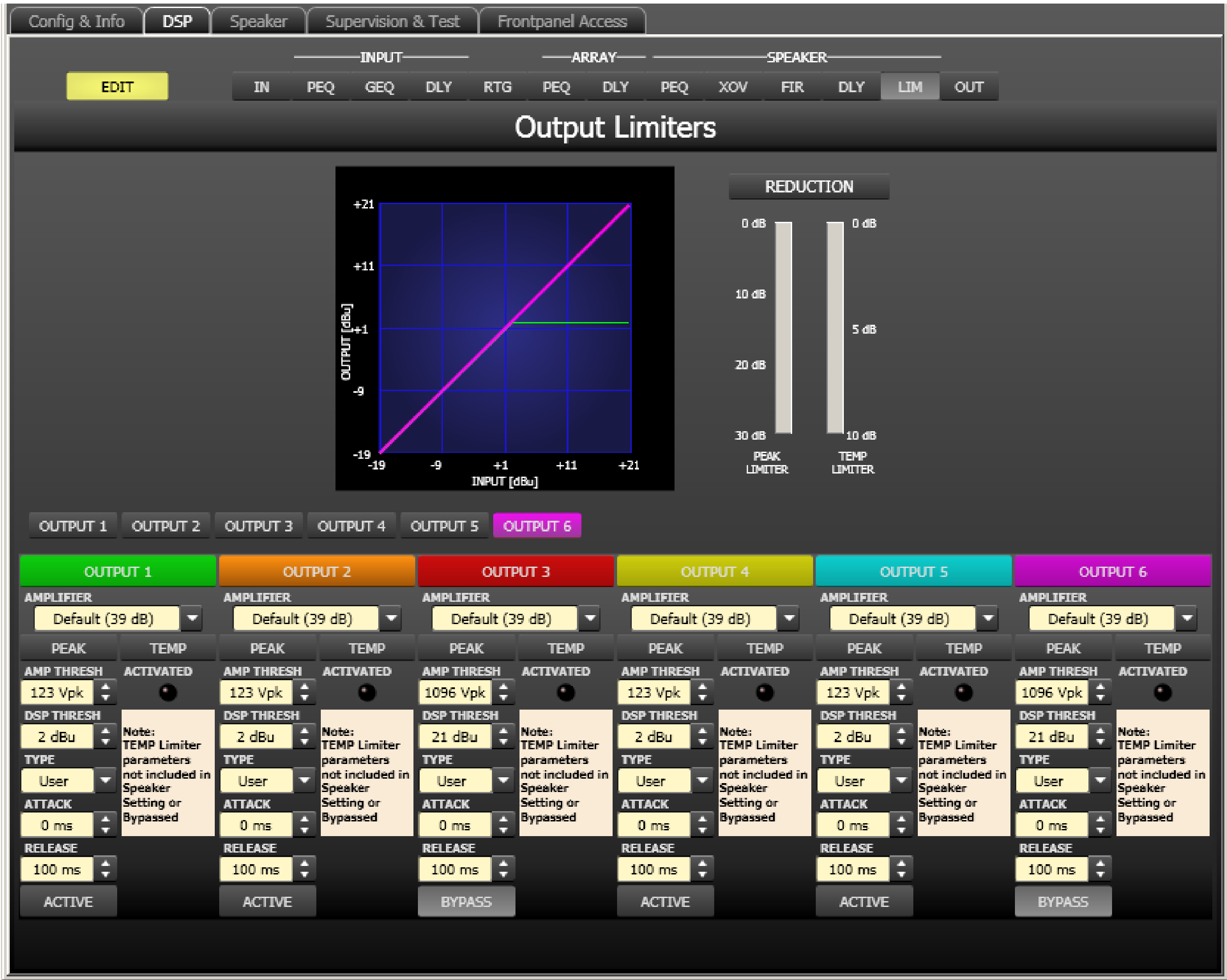
General Parameters

Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display
The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT LIMITERS

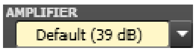
Each output channel of the sound system processor offers a peak limiter and a TEMP limiter. These functions can be accessed via the Output Limiters window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.
Clicking on the 12. block (LIM) in the Flow Diagram Selector or double clicking on the SPEAKER PROCESSING LIMITERS block in the large flow diagram opens the Output Limiters window.



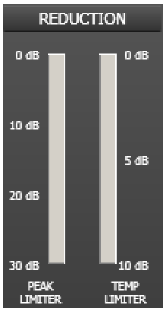
Channel Selection

Element	Description
	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for limiter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all limiter settings of the corresponding output to any other limiter within the same project.

Limiter Parameters

Element	Default	Range	Description
	Default (39 dB)	Default (39 dB), S900, S1200, CL800, CL1200, CL1600, CL2000, LX1600, LX2200, LX3000, L1000 (0dBu), L1000 (+6dBu), L1000 (26dB), L1600 (0dBu), L1600 (+6dBu), L1600 (26dB), L2400 (0dBu), L2400 (+6dBu), L2400 (26dB), H 2500 (0dBu), H 2500 (32dB), H 2500 (35 dB), H 5000 (0dBu), H 5000 (32dB), H 5000 (35dB), SL900, SL1200, SL1800, SL2400, DSA8204, DSA8206, DSA8209, DSA8212	Select the amplifier type connected to output of the DSP 600.
	123 Vpk		AMP THRESH determines the audio signal level above which the peak limiter starts operating.
	2 dBu		DSP THRESH determines the audio signal level above which the peak limiter starts operating. This value may change depending on the AMP- LIFIER type that is selected, as the sensitivity and output power are automatically calculated with the Vpk value to provide DSP Threshold.
	User	User, Hi, Mid, Lo, Sub	TYPE allows a user to select a bandpass type, and the software will enter appropriate default time constants for the bandpass selected. DYNACORD speaker settings include factory-defined time constants, so this section is only for use when creating DSP settings from scratch.
	0 ms	0 to 50 ms	ATTACK determines how fast the limiter reduces amplification when the threshold is exceeded.
	100 ms	10 bis 1000 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
			Press the ACTIVE button to deactivate the peak limiter.
			The ACTIVATED LED lights green if the TEMP limiter is active.

Gain Reduction Meters

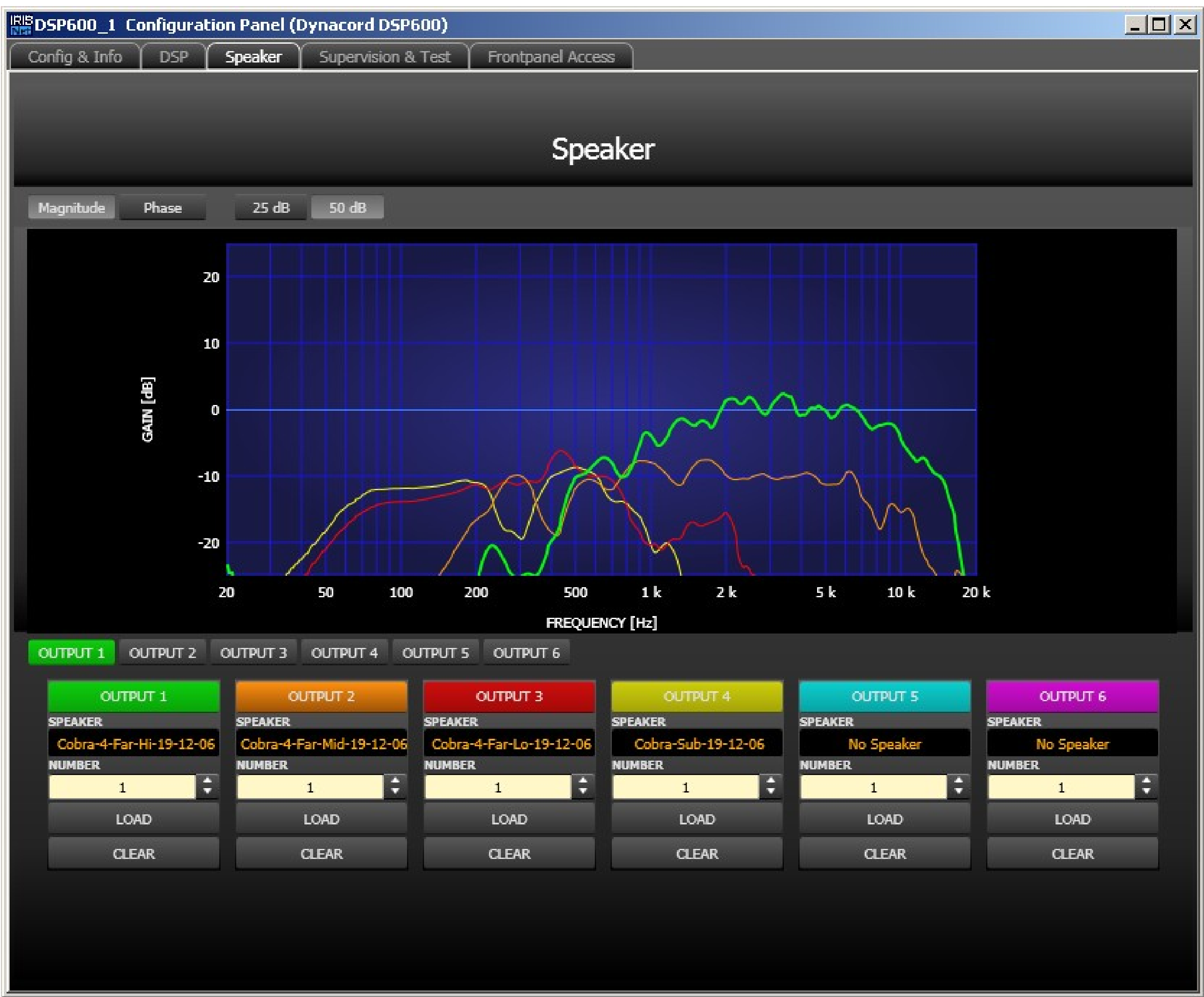
Element	Description
	These indicators shows the reduction in dB that is applied to the audio signal by the Peak limiter (PEAK) or the TEMP limiter (TEMP LIMITER). Level reduction is indicated as vertical yellow bar graph.

Editing Limiter Parameters by ”Dragging the Mouse” in the Graphics Display

Active limiters (BYPASS button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding limiter by vertically dragging the mouse.

Speaker

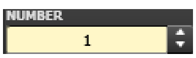
The Speaker Dialog offers the possibility to load the acoustic measurement data of different loudspeaker systems, assign it to the sound system processor channels and display the acoustic results. The speaker system datasets, which are provided as “speaker files” (*.spk), contain factory-measured frequency- and phase responses of loudspeaker systems.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the sound system processor. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of sound system processor plus speaker transfer functions.
Clicking on the Speaker tab in the Configuration Panel opens the Speaker Dialog.



Indication on the Graphic Display

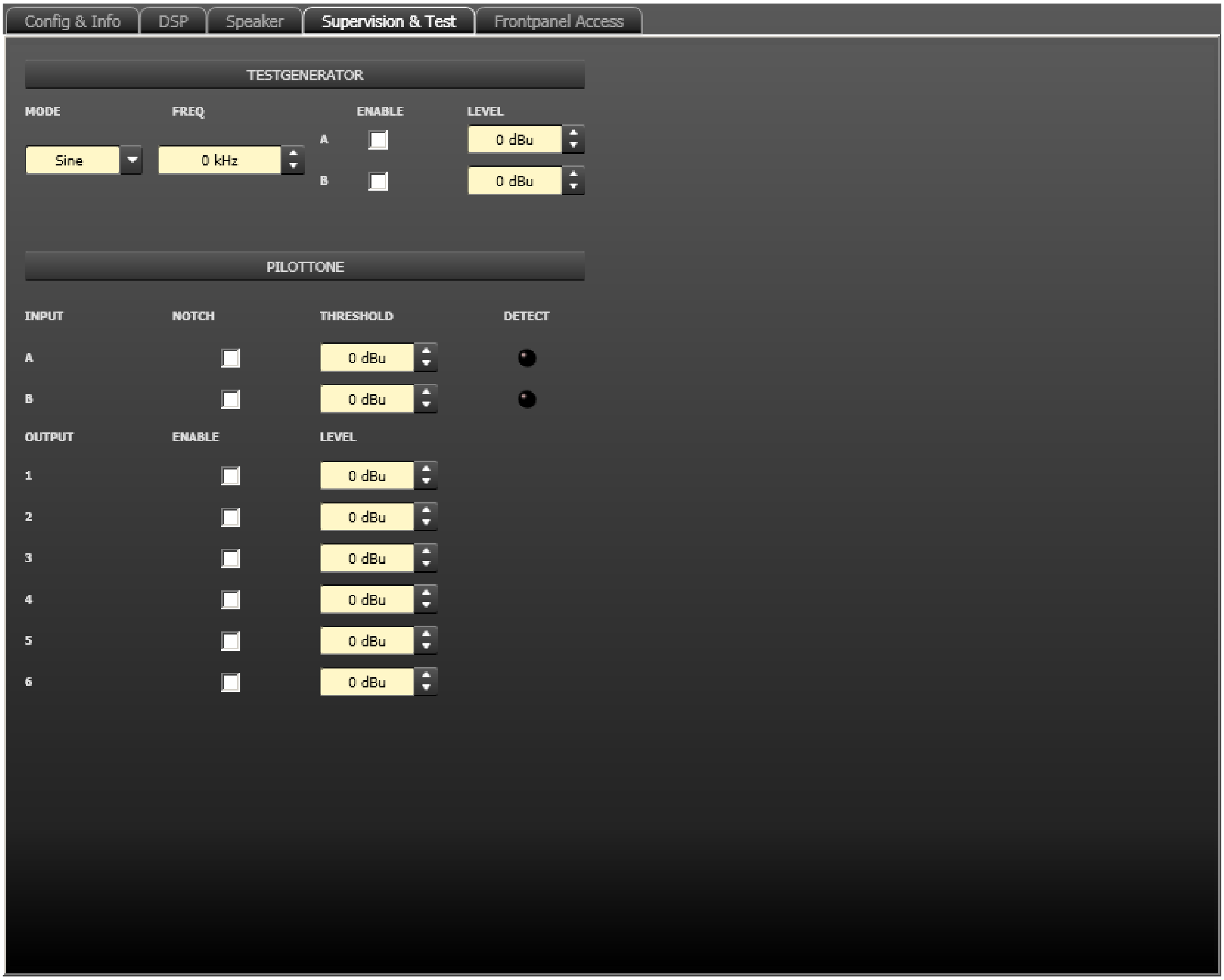
Element	Description
	Switch for toggling between frequency response (magnitude) and phase response (phase) display
	Switch for adjusting the scale of the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)

Channel Parameters

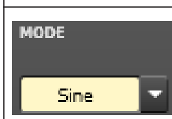
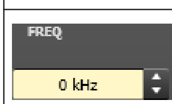
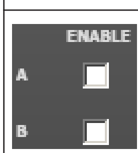
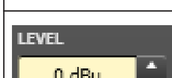
Element	Default	Range	Description
			Switch for selecting output 1, 2, 3, 4. 5 or output 6 for limiter editing.
			The name of the loaded loudspeaker model is shown in the black-shaded field.
	1	1 to 10	The NUMBER parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel.
			Clicking the LOAD button opens a dialog that allows the selection of the desired speaker file.
			Clicking the CLEAR button clears the previously loaded measured speaker data of the selected channel.

Supervision & Test

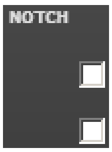
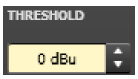
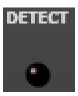
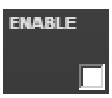
The Supervision window allows configuration of the test and pilot tone generator, additionally the status of the pilot tone detection is indicated.



Test generator

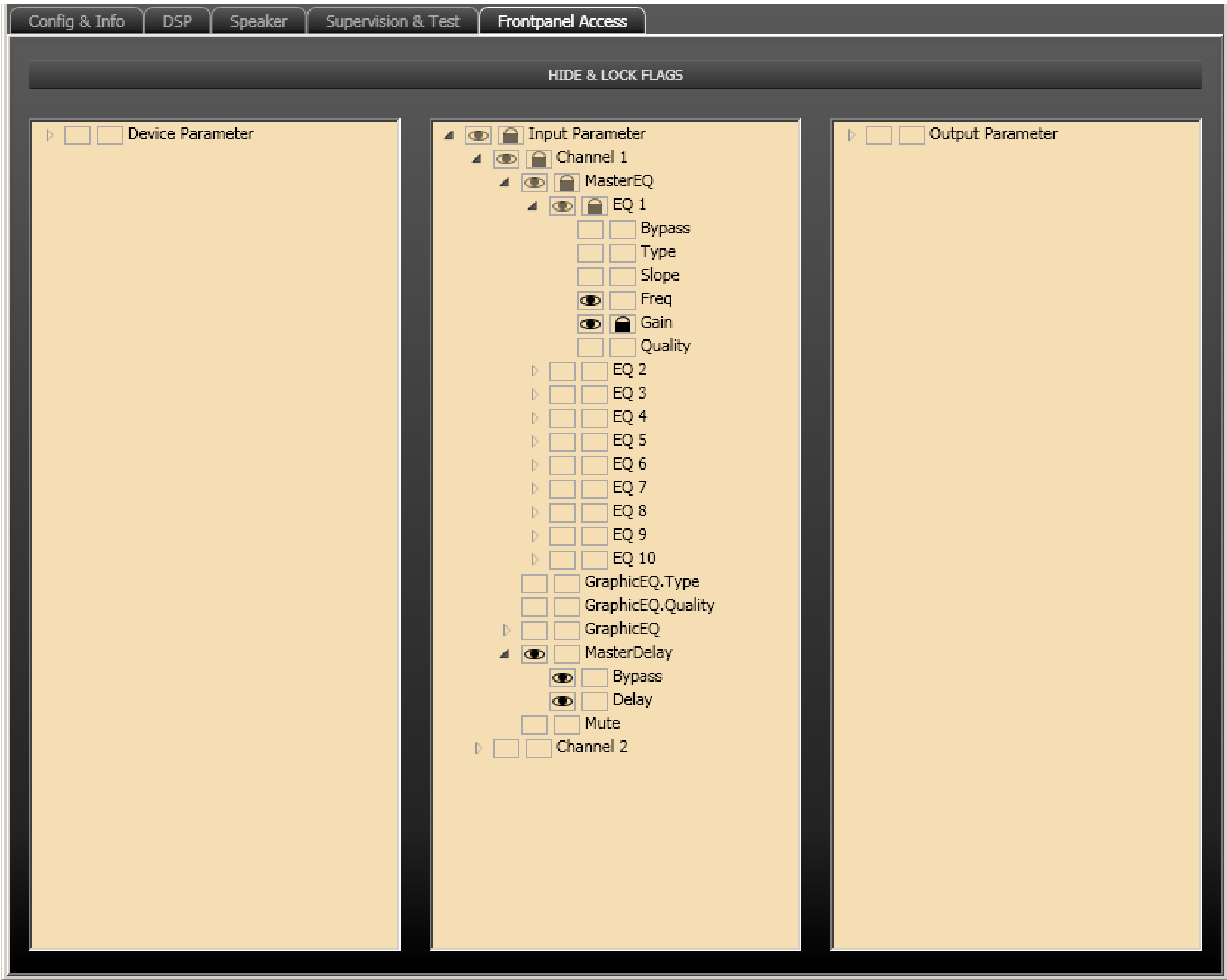
Element	Default	Range	Description
	Sine	Sine, White Noise, Pink Noise	Select the mode of the test generator.
	0 kHz	20 Hz to 20 kHz	Set the frequency of the generated sine signal. The FREQ parameter is only available is MODE Sine is selected.
	Off	On / Off	Activate the test generator for input channel A and/or input channel B.
	0 dBu	-80 to 0 dBu	Enter the signal level for input channel A or B in dBu.

Pilot tone

Element	Default	Range	Description
	Off	On / Off	The checkbox activates a notch filter in input A or B. The notch filter filters an existing pilot tone out of the input signal.
	0 dBu	-80 to 0 dBu	This field sets the pilot tone detection’s threshold.
			The pilot tone detection results in OK (LED lights green) when the level of the pilot signal exceeds the threshold. Without a pilot tone being present or if the signal level is below the set threshold, analysis results in a fault message on the corresponding input channel (LED does not light).
	Off	On / Off	Checkbox for activating or deactivating the pilot tone generator.
	0 dBu	-128 bis 0 dBu	Enter the pilot tone signal level in dBu.

Frontpanel Access

This dialog allows selecting which parameters should be visible and/or editable on the front panel of the DSP 600. By default all parameters are visible (eye icon set) and can be edited (lock icon not set). Remove the eye icon of a parameter that should not be visible on the front panel. Activate the lock icon of a parameter that should not be editable at the front panel.


Config & Info
DSP
Speaker
Supervision & Test
Frontpanel Access
HIDE & LOCK FLAGS
Device Parameter
Input Parameter
Channel 1
MasterEQ
EQ 1
Bypass
Type
Slope
Freq
Gain
Quality
EQ 2
EQ 3
EQ 4
EQ 5
EQ 6
EQ 7
EQ 8
EQ 9
EQ 10
GraphicEQ.Type
GraphicEQ.Quality
GraphicEQ
MasterDelay
Bypass
Delay
Mute
Channel 2
Output Parameter

Dx46 FIR-DRIVE

The Electro-Voice Dx46 Digital System Processor is a universal two-input, six-output digital signal processor with the configuration flexibility to handle a multitude of audio system needs and applications; installed sound, house of worship, convention & meeting facilities, concert touring, club, portable sound reinforcement and more. The internal signal processing structure can be configured as 2-way stereo + full-range, 3-way stereo, 4-way mono + fullrange, 5-way mono + full range, 3-way stereo with a mono sub + full-range, 4-way stereo with mono sub and low frequency and finally as a freely assignable 2 x 6 matrix router.
The Dx46 replaces entire racks of signal processors previously needed to properly configure and control sound reinforcement systems with a single Dual-Core DSP processor. The substantial advantages of the Dx46 over discrete signal processing racks include:

– 24-bit, 48 kHz digital signal path
– No patch cables to fail or add noise
– Optimal gain structure throughout all stages of signal processing; no gain matching from processor to processor
– Recallable factory and user presets; instant system reconfiguration for differing applications and performances
– Easy, intuitive operation and editing with a PC and IRIS-Net

FIR-DRIVE

The Dx46 includes Finite Impulse Response (FIR) filters at each output for loudspeaker linearization. Using FIR filters has the following advantages, compared to using IIR filters (e.g. Bessel, Butterworth,...).:

– extremely linear frequency response
– very high stop-band attenuation
– linear phase systems

To sum up, FIR-DRIVE allows the linearization of frequency and phase of your Electro-Voice loudspeakers. Activating FIRDRIVE is as easy as loading a FIR Speaker Setting in the output channel of the Dx46. The IRIS-Net software is used for loading Speaker Settings, and lots of Electro-Voice FIR Speaker Settings are included. Please refer to the documentation of IRIS-Net for more details about using Speaker Settings.
Each Dx46 Digital System Processor includes the following signal processing blocks:

INPUTS

– Pilot tone detection
– VU Metering of input signal
– Analog or digital (AES/EBU) Inputs
– 24-bit, 48 kHz A/D converters
– 10-band parametric equalizer
– 31-band graphic equalizer
– Delay

MATRIX ROUTER / MIXER

– Two inputs (stereo)
– Summed left / right (mono) input
– Six assignable outputs

OUTPUTS (EACH)

– Array control (5-band equalizer +delay)
– Cross-over (hi-pass / low-pass filters), with selectable filter types
– 6-band parametric equalizer
– FIR filter with 512 Taps

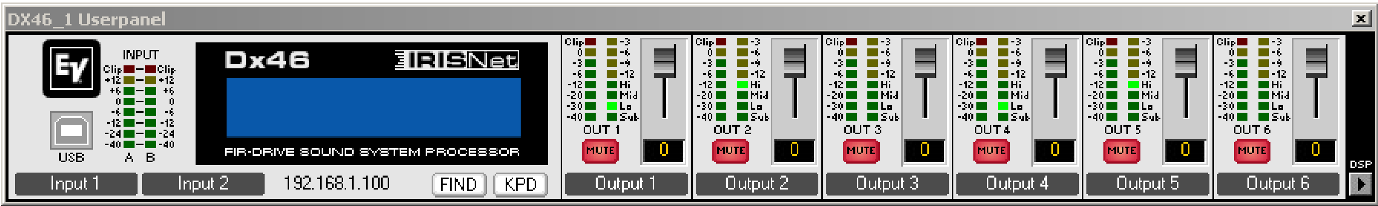
- Delay
- Polarity
- Look-ahead Peak limiter with Peak RMS detection
- TEMP Limiter for long-term loudspeaker protection
- Level & Mute
- 24-bit, 48kHz D/A converters
- Pilot tone generator
- VU Metering
- Output assignment display LEDs; sub, low, mid & high
- Mute button
- Gain reduction meters

ADDITIONAL FEATURES INCLUDE:
- Electronically balanced XLR inputs and outputs
- XLR thru connectors (analog + AES/EBU)
- -6 dB switchable input level PAD
- Test generator (Sine, pink noise, white noise)
- Contact closure interface
- USB port (front) and Ethernet port (rear) for connection to PC with IRIS-Net software; preset editing and real time parameter control and monitoring.
- Firmware updates via USB port or Ethernet port
- FLASH memory for preset storage and firmware upgrades
- 192 x 32 back-light graphic LCD display
- LCD navigation / editing controls
- DSP block direct access controls
- Auto-ranging internal power supply; 100-240 V AC, 50-60 Hz
- Standard IEC A.C. inlet with external, replaceable fuse

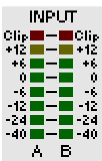
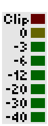
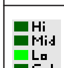
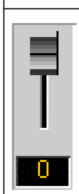
Dx46 Device
Start by creating an Dx46 Device in your IRIS-Net project. Drag an Dx46 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the Ctrl key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an Dx46 device icon opens the User panel.

Dx46 Userpanel

The Dx46 User panel provides access to the controls and indications on the Dx46 front panel.



Indications and Functions of the Dx46 Userpanel

Element	Description
	The level meter displays are meant for visual monitoring of the input signal levels, individually showing the peak value of the correspondent input signal in dBu. The input control should be set to a position so that the meter instruments indicate a level between -6 and -12 dB. To prevent internal clipping, make sure that the CLIP LEDs are not lit.
	In online mode the LCD display is identical on the Dx46 User panel and the device.
	These LEDs indicate the peak level of the corresponding outputs. The level is shown as headroom relative to the D/A clip or limiter threshold, as selected in the Dx46 menu. The Dx46 should be operated in a range, so that the clip-LEDs are not lit. Otherwise, this could lead to internal clipping.
	Each output channel has a four-segment gain reduction meter that shows the gain reduction of the output channel Limiter on output signal; from -3dB to -12dB.
	Each output channel has a four-segment function display for informational purposes only. For any given configuration possible with the Dx46, an output channel may be identified as a sub, low, low/mid, mid, mid/hi, hi or full range output. One or two adjacent LEDs are displayed to indicate all possible output band-passes. (Full range is indicated by no lit LED's.)
	Each output channel has a lighted Mute button. Pressing the Mute button turns off the output of that channel. The button lights red as an alert. Press the Mute button again to restore the output channel's signal.
	These controls are used to set the output levels of channels 1 to 6, allowing the matching of the Dx46 to the input levels of the devices chained in sequence. Correctly setting these controls results in an improved S/N ratio. The digital output gain control should be used when higher output levels are needed. Use the controls to attenuate the output levels. It is not recommended to use the digital output gain control for massive attenuation, since this would reduce the dynamic range of the D/A-converters.
	Label of input or output channels, the labels can be edited at the Config & Info window.

	Clicking on the DSP button opens the Configuration Panel, which provides access to all DSP and speaker parameters.
192.168.1.100	Indicates the IP address of the Dx46’s Ethernet port (factory setting: 192.168.1.100). Click to edit the address.
FIND	A click on the FIND button lets the LEDs on the front panel of the Dx46 blink. When on-line, this allows for easy identification of which Dx46 the user is currently communicating with. Click on the FIND button again to stop the LEDs blinking.
KPD	A click on the KPD button opens the Keypad dialog. When on-line, the buttons in the Keypad dialog have the same function as the buttons on the front panel.

Keypad

Configuration Panel

Clicking on the „SET“ soft key in the Dx46 Userpanel or selecting the entry „Dx46 UI Dialog“ from the context menu of the device opens the Configuration Panel. The Configuration Panel allows configuration of all Dx46 parameters. It also provides access to different test functions. The window is divided into several pages according to the corresponding function groups:

Dialog	**Description**
Config & Info	This page provides information about the Dx46 and allows making several basic settings as well as programming control functions.
DSP	The DSP page provides an overview plus access to all DSP functions (Input, Array and Speaker) of the Dx46.
Speaker	This page allows the loading and displaying speaker data.
Supervision & Test	This page provides access to several settings for test generator and pilot tone detection.
Frontpanel Access	This page allows selection of which parameters should be visible or editable from the front panel.

Config & Info

The Config & Info window provides information and basic settings for the selected Dx46. Additionally, editing labels and configuration of control port functions is possible as well.

Device Info

Element	Description
DEVICE MODEL	Shows the signal processor type
FIRMWARE	Shows the software‘s version number.
UPTIME	Shows the uptime of the Dx46.

Network Settings

Element	Default	Description
IP ADDRESS	192.168.1.100	IP address of the Dx46
MAC ADDRESS		MAC address of the Dx46

Master/Slave Settings

Element	Default	Description
MODE	off	The master/slave settings only work if more than one Dx46 device is connected to a network. Devices that are Master or Slave have always idential parameter settings. Select „Master“, if this Dx46 should write the parameter settings to one or more other Dx46 (Slave). Select „Slave“, if the parameter settings of this Dx46 should be read from another Dx46 (master). Select „off“, if the parameter settings of this Dx46 should be independent from other devices.
NETWORK ID	0	Each master Dx46 connected to the network must have an unique network id. Enter the id of the Master Dx46 the parameters should be read from if this Dx46 is used as „Slave“. Multiple Dx46s can be Slaves to a single Master, if desired.
	all groups selected	If the MODE „Slave“ is selected, choose the parameter groups that this Dx46 should read from the Master Dx46. Following groups are available: – D: Parameters of the device – A or B: Parameters of input A or B – 1 to 6: Parameters of output 1 to 6

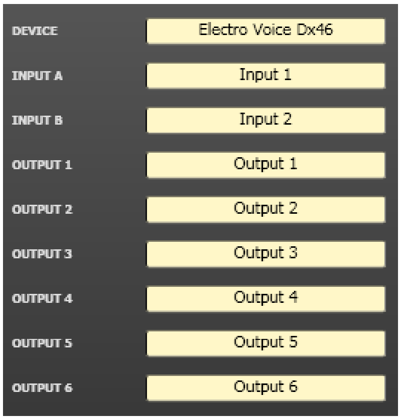
Input Settings

Element	Default	Range	Description
MODE	Analog	Analog, AES/EBU	Select the analog or the digital (AES/EBU) audio inputs of the Dx46.
INPUT GAIN	0 dB	-60 to +12 dB	Adjust the input gain of the audio input.
ADC OVFL			The LED lights red for 2 seconds if there is an overflow of the A/D converter.
-6dB PAD	off	on, off	Input levels to the Dx46 can be reduced 6dB prior to the A/D converter to compensate for higher- level output from mixers and other audio devices. For ideal signal to noise performance when con- necting the Dx46 to high output level devices engage the 6dB PAD („on“) rather than turning down the output of the connected device.

Digital Input

Element	Default	Range	Description
LOCKED, ERROR			If the LOCKED LED lights green, the input is synchronized to the incoming signal and the audio is correctly transmitted. When signal transmission fails, the ERROR LED lights red.
AES/EBU SAMPLE RATE	-	32 to 192 kHz	Shows the sampling rate of the incoming signal when the input has been successfully synchro- nized.

Device & Channel Labels

Element	Description
	The labels of the Dx46 and its input and output channels are shown in a clear structure. All labels can be edited. Changes are immediately reflected in the different panels and windows (Userpanel, flow diagram). The DEVICE label is indicated on the display at the Dx46 front panel. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

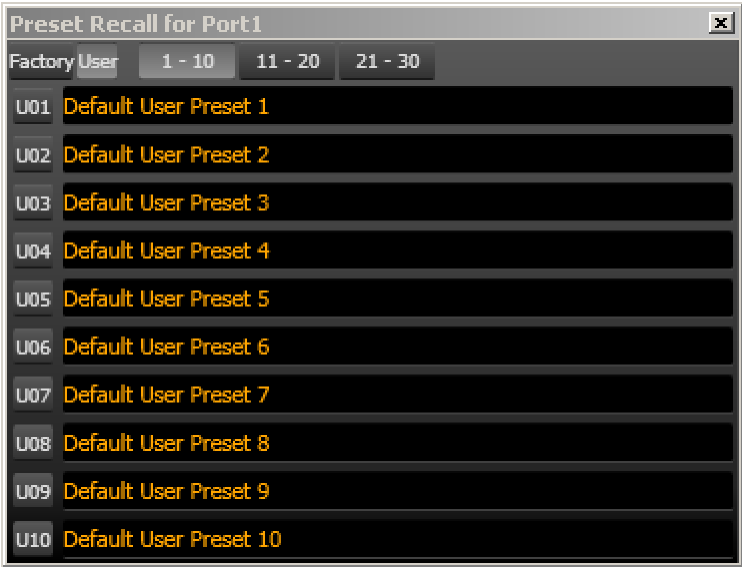
Control Port

The Control Port of the Dx46 provides five control inputs and a reference connection for ground. The control inputs can be used for the recall of presets. For more information and electrical specifications of the control port, please refer to the Dx46 manuals.

Element	Description
	Description and current state of the input. The LED lights green if the input is connected to ground.
	Description of the preset to be recalled by the input.
	Clears the preset assignment to the input.
	Opens the Preset Recall for Port x dialog. This dialog allows assignment of a factory or user preset to the input.

Preset Recall for Port x

This dialog lists the 60 Factory Presets and 30 User Presets of the Dx46.



Element	Description
Factory / User	Switch between the Factory Presets or User Presets.
1-10, 11-20, 21-30, ...	Select the preset group that should be listed.
U01...U10	Press the button of the preset, that should be assigned to the input.

DSP
The DSP pages provide overview and access to all DSP parameters of the sound system processor. Within this window you can use the Flow Diagram Selector to link to different function groups.

Element	Description
	The current preset is in Edit mode if the yellow EDIT button is indicated. Pressing the EDIT button „compares“ the edited preset, if parameters have been altered, to the original unedited preset. This compare function will audibly switch between the altered parameters and the previously stored settings, allowing you to hear the effect of any DSP changes that have been made. Use this feature to monitor progress in editing or creating presets. Subsequently recalling a new preset will prompt you to save changes, which you may do or not.

FLOW DIAGRAM SELECTOR
The Flow Diagram Selector can be accessed from any DSP page offering navigation means within the DSP signal processing functions. The Flow Diagram Selector lets you select different function blocks, where the actually selected block is displayed in a light grey engaged field.

A short description of each DSP page is provided in the following table. Please refer to the corresponding chapters for a more detailed explanation.

Element	Description
Flow Diagram	The signal flow display provides an overview of the DSP settings. This area also includes all controls for the preset location/file management and configuration settings.
Input Parametric EQ	The Input Parametric EQ page provides access to the two 10-band parametric equalizers of the sound system processor inputs.
Input Graphic EQ	The Input Graphic EQ page provides access to the two 31-band graphic equalizers of the sound system processor inputs.
Input Delay	This page allows the programming of delay lines for the input channels A and B.
Array Parametric EQ	The Array Parametric EQ page offers access to the 5-band parametric equalizers of the sound system processor outputs.
Array Delay	This page allows the programming of delay lines for the output channels.
Output Parametric EQ	The Output Parametric EQ page offers access to the 6-band parametric equalizers of the sound system processor outputs.
Output X-Over	Frequency crossover-filters as well as the parameters gain and polarity for all output channels are located in the Output X-Over area.
Output FIR	This page provides a FIR-Filter for each output channel.
Output Delay	This page allows the programming of delay lines for the output channels.
Output Limiters	This page provides access to Peak limiter and TEMP limiter of each output channel.

FLOW DIAGRAM

The Flow Diagram window shows a signal flow diagram, which offers a quick overview of all DSP setting of the Dx46.

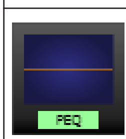
- Output muting,
- routing channels,
- setting output level,
- editing configuration LEDs (Free Configuration mode only),
- import and export of Speaker Settings

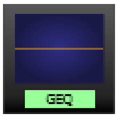
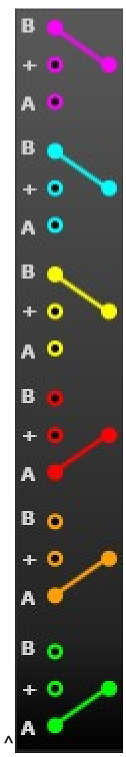
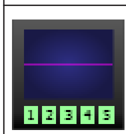
can be done directly in the diagram. Clicking onto the corresponding function blocks lets you access all other DSP parameters. All parameters that are necessary for the saving, loading and previewing of presets are also accessible from this window.

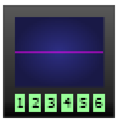
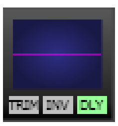
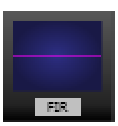
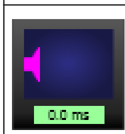
The FLOW DIAGRAM window opens when clicking on the first (IN), fifth (RTG) or 13. block (OUT) in the Flow Diagram Selector.

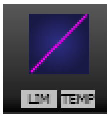
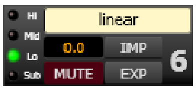
Function blocks

The caption below the function blocks is shown in green if the function, or at least one filter of the block, is activated.

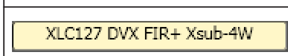
Element	Description
	INPUT PEQ Block: The INPUT PEQ block displays the 10 EQs of the corresponding input channel. The graph shows the frequency response of the EQ block. A single click with the left mouse button onto this block opens the Input Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding EQ block to any other Dx46 INPUT PEQ block within the same project.

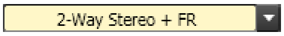
	INPUT GEQ Block: The INPUT GEQ block displays the 31 graphical EQs of the corresponding input channel. The graph shows the frequency response of the GEQ block. A single click with the left mouse button onto this block opens the Input Graphic EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding GEQ block to any other Dx46 GEQ block within the same project.
	INPUT DELAY Block: This displays the Delay of the input channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Input Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other Input Delay block within the same project.
	ROUTING Block: Here you can assign the output channel routing. The circles next to A and B allow selecting the input signal for the corresponding output channel. The circle next to the + allows selecting the summed input signal for the corresponding output channel.
	ARRAY PEQ Block: The ARRAY PEQ block displays the 5 Array EQs of the corresponding output channel. The 5 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Array Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding ARRAY PEQ block to any other EQ block within the same project.

	ARRAY DELAY Block: This displays the Array Delay of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Array Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other ARRAY DELAY block within the same project.
	SPEAKER PROCESSING PEQ Block: The SPEAKER PROCESSING PEQ block displays the 6 Channel EQs of the corresponding output channel. The 6 LEDs indicate which EQ-bands are being used while the graph shows the frequency response of the PEQ block. A single click with the left mouse button onto this block opens the Output Parametric EQ page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Speaker EQ block to any other EQ block within the same project.
	SPEAKER PROCESSING X-OVER Block: This block represents the crossover within the corresponding output channel. The graph shows the frequency response that results from the set X-Over parameters. Three additional LEDs indicate the status of gain trim (TRIM), polarity (INV) and delay (DLY). A single click with the left mouse button onto this block opens the Output X-Over page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding X-Over block to any other Dx46 X-Over block within the same project.
	SPEAKER PROCESSING FIR FILTER Block: This block represents the FIR Filter within the corresponding output channel. The graph shows the frequency response that results from the set FIR parameters. The LED indicate if the FIR Filter is being used. A single click with the left mouse button onto this block opens the Output FIR page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding FIR Filter block to any other FIR Filter block within the same project.
	SPEAKER PROCESSING DELAY Block: This displays the SPEAKER PROCESSING DELAY of the output channels. The delay-value is displayed together with the measurement unit. The graph shows the approximate usage of delay memory capacity. A single click with the left mouse button onto this block opens the Speaker Processing Delay page. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Delay block to any other Dx46 Speaker Delay block within the same project.

	SPEAKER PROCESSING LIMITERS Block: This block provides graphical display of the limiter functions of the corresponding output. The two LEDs indicate whether peak limiter or TEMP limiter have been activated. The graph provides indication of the set values. Clicking with the right mouse button opens the Copy & Paste menu, which allows copying all parameters of the corresponding Limiters block to any other Dx46 Limiters block within the same project.
	Output Block: For any given configuration possible with the Dx46, an output channel may be identified as a sub, low, low/mid, mid, mid/hi, hi or full range out- put. One or two adjacent LED are displayed to indicate all possible output bandpasses. (Full range is indicated by no lit LED’s.) When Free Configuration is selected the LEDs can be set manually. In online mode the LEDs here and at the front panel are identical. A click with the right mouse button onto OUT 1 to OUT 6 opens the Copy & Paste menu, which allows copying of all parameters of the corresponding output channel to any other Dx46 output channel within the same project. The numerical field is identical to the numerical field below the level controls in the Userpanel, click to edit the value. The MUTE button is for attenuating the output level of the corresponding output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute- function and the output is again active. The MUTE button is virtually disengaged and not lit. The IMP or EXP buttons allow import and export of Speaker Settings. A Speaker Settings file contains the loudspeaker specific settings for the SPEAKER PROCESSING blocks. The text field allows editing the description of the Speaker Setting file to be exported. Importing a Speaker Setting automatically imports the corresponding Speaker File.

Status Indication

Element	Description
	Displays the number according to the actually audible preset. However, this is only true if the EDITED LED lights green, i.e. no DSP parameter has been changed since the last RECALL.
	Indicates the name of the actually audible preset. Click to edit the preset name.
	The EDITED indicator provides information whether a parameter has been altered since the last RECALL. If the indicator lights red, parameters have been edited and therefore differ from the ones of the preset that is shown.

	The Dx46 offers 7 pre-defined configurations. A configuration is a basic setting that includes the routing of inputs and outputs, the function of the outputs (Sub, Lo, Mid, Hi, Fullrange), as well as basic X-Over parameters. If you do not want to use one of the pre- defined configurations, you have also the possibility to operate the Dx46 in Free Configuration mode. This mode offers access to all parameters and basically any input/output routing can be programmed. Additionally, the output assignment (output function - Sub, Lo, Mid, Hi, Fullrange) is also freely definable.
	The Dx46 defaults to Standard Edit mode wherein input and output channel parameters are appropriately linked. In Full Edit mode, no parameter links are enforced, regardless of the configuration selected. When switching from Full Edit mode to Standard Edit mode, parameters are linked again. At the same time parameters settings in linked channels (upper numbers) will be overwritten.

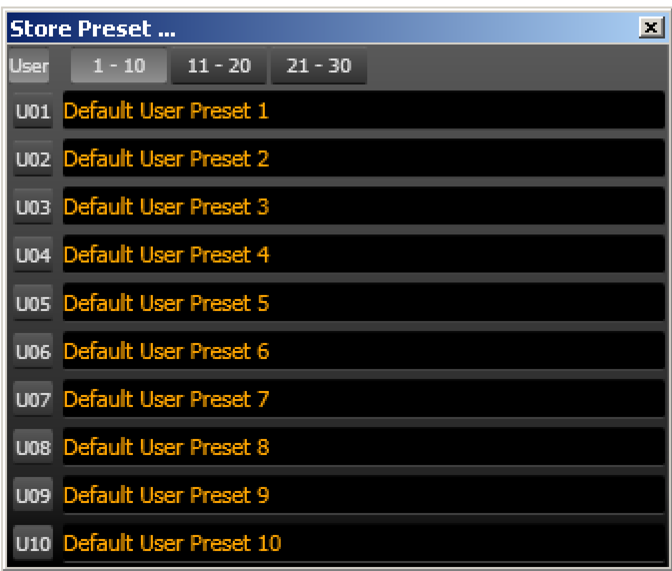
Recall a Preset

Element	Description
	Clicking the RECALL... button opens the Recall Preset dialog, where you can select and recall a Preset.

Caution!
The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!
Consequences

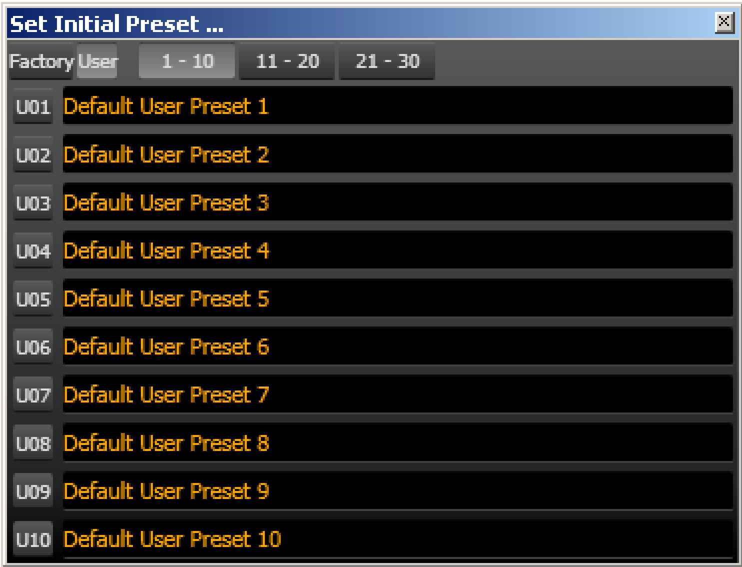
Store a Preset

Element	Description
STORE	STORE saves all currently set DSP parameters together with the entered name into the current preset. Store is only possible if a user program number is selected.
STORE TO ...	Clicking the STORE TO.... button opens the Store Preset... dialog. In this dialog the Program Number can be selected.



Setting a Startup presets

Element	Description
STARTUP PRESET U01	The indicated preset is loaded after power on or reset of the Dx46.
ASSIGN ..	Clicking the ASSIGN .. button opens the Set Initial Preset... dialog. In this dialog a factory preset or user preset can be selected as startup preset.
X	Clicking the X buttons clears the Startup preset selection.



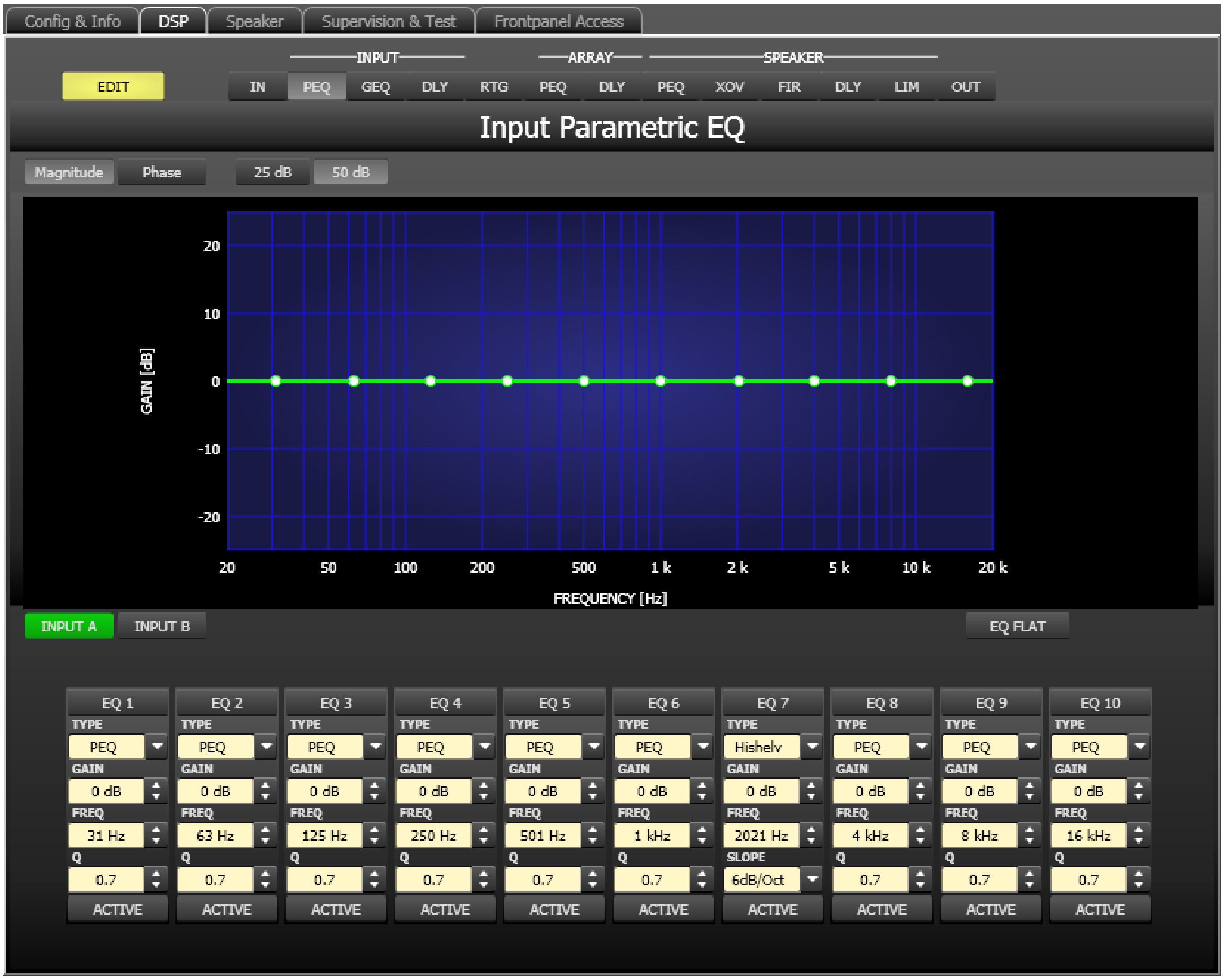
Import / Export a Preset

IRIS-Net allows the storing of all DSP parameters of an sound system processor together with the according preset name in a file on a PC, and to load sound system processor parameters from these files. Therefore, IRIS-Net creates a sub- directory \Presets during installation, where all factory-presets are saved in to. It is recommended to save your own presets in this directory as well. For improved organization, creating more sub-directories within the directory \Presets is permissible.

Element	**Description**
IMPORT PRESET	After clicking onto IMPORT PRESET an ”Open File ...” dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired preset file to be opened. This loads and afterwards displays all DSP parameters that are stored within that file. **CAUTION: The loaded preset becomes instantly audible when in on-line mode. Be sure to select the desired preset with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT PRESET	After clicking onto EXPORT PRESET a ”Save File ...” dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click the SAVE button to store all DSP parameters together with the corresponding file name. ”.ds” is automatically added as file extension.

INPUT PARAMETRIC EQ

Both input channels of the sound system processor employ 10-band parametric equalizers each, which allow programming highly variable speaker equalization to match a PA-system to different environmental and acoustical requirements.
The Master-EQ is selected by clicking on the second block (PEQ) of the flow diagram selector or on the INPUT PROCESSING PEQ block in the full-scale flow diagram.



Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

Element	Description
	Switch for selecting input A or input B for filter editing and display. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other Dx46 Input PEQ filter bank within the same project.

Filter Parameters

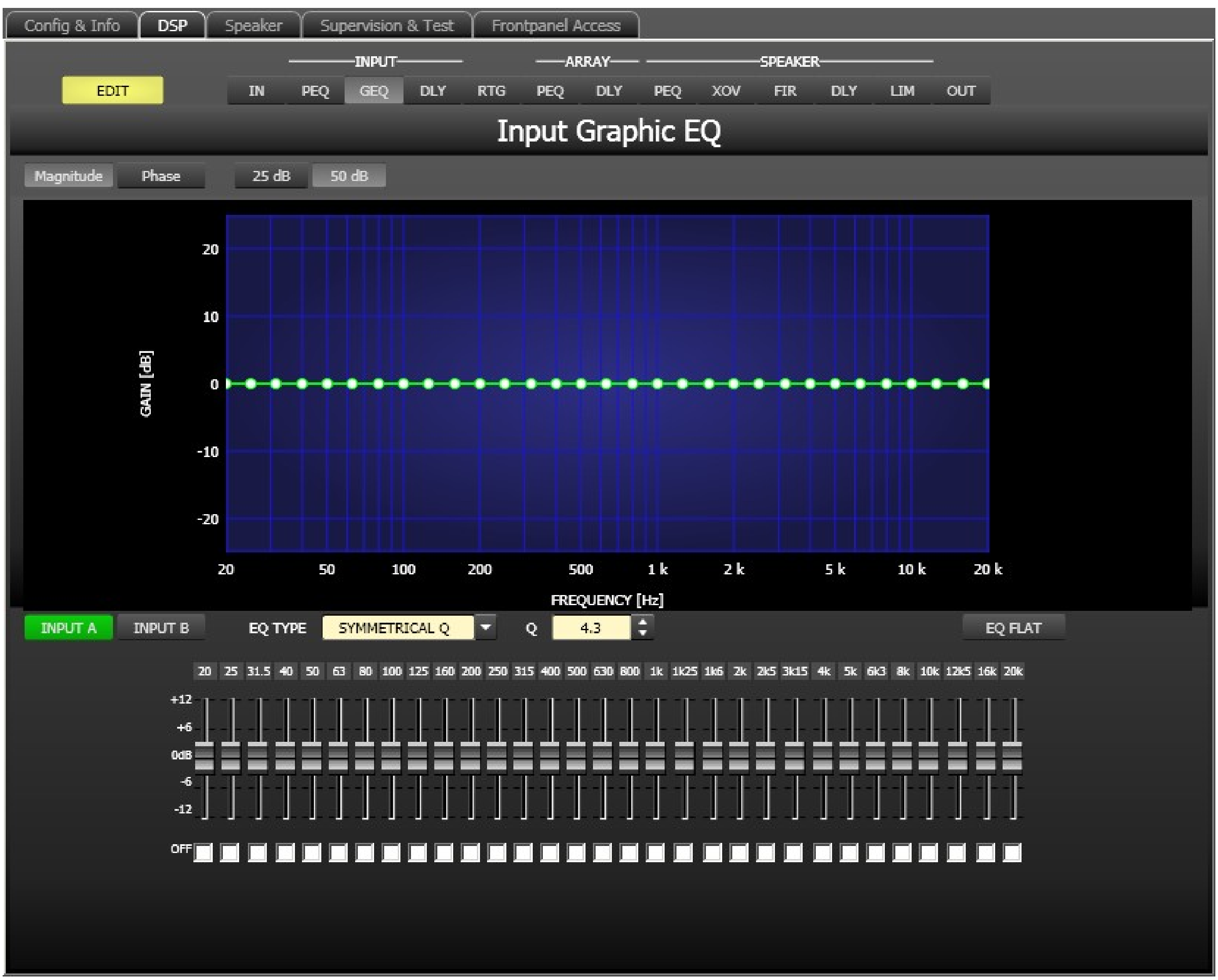
Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	31 / 63 / 125 / 250 / 500 / 1k / 2k / 4k / 8k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/ Lopass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
GAIN 0 dB	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
ACTIVE			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
EQ FLAT			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via “Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its magnitude (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values of parametric EQs and Hipass/Lopass filters.
For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

INPUT GRAPHIC EQ

Both input channels employ 31-band graphic equalizers each, which allow programming highly variable speaker equalization to match a PA-system to different environmental and acoustical requirements.
The Master-EQ is selected by clicking on the third block (GEQ) of the flow diagram selector or on the INPUT PROCESSING GEQ block in the full-scale flow diagram.



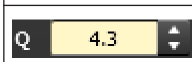
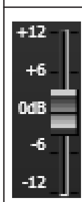
Graphics Display Indication

Element	Description
	Switches between frequency (magnitude) and phase response (phase) indication
	Switch for selecting dB-axis scaling of 25 dB (± 12.5 dB) or 50 dB (± 25 dB)

Channel Selection

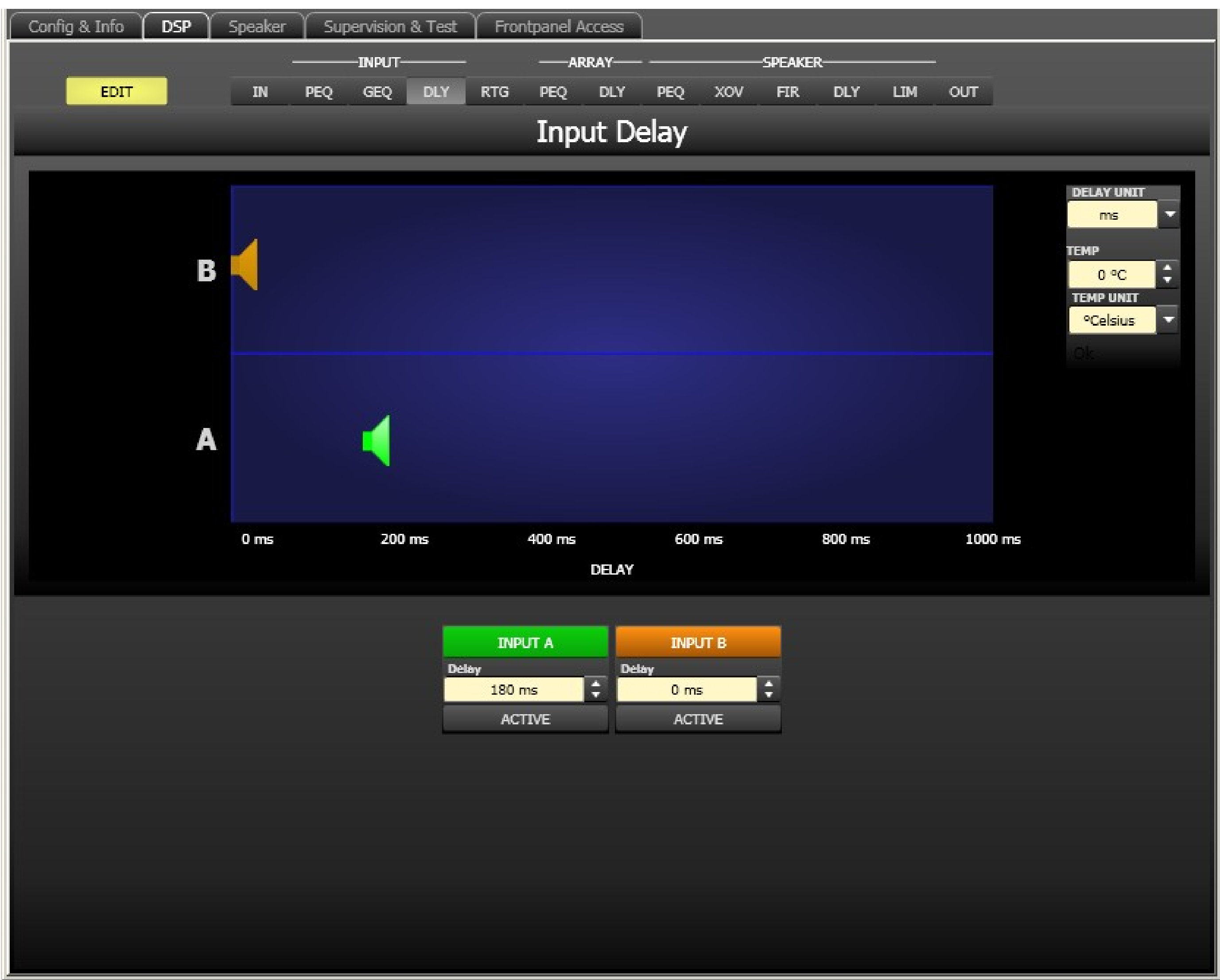
Element	Description
	Switch for selecting input A or input B for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other Dx46 Graphic EQ filter bank within the same project.

Filter Parameters

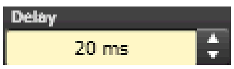
Element	Default	Range	Description
	SYMMETRICAL Q	SYMMETRICAL Q, PROPORTIONAL Q, CONSTANT Q	Switching the graphic equalizer’s type between SYMMETRICAL Q, PROPORTIONAL Q and CONSTANT Q. SYMMETRICAL Q: The filters have an identical Q at all accentuation settings. The lowering frequency responses are symmetrical to the accentuation frequency responses. PROPORTIONAL Q: A Filter’s Q increases as soon as accentuation or lowering of the filter increases, with the effect that the equalizer becomes ”sharper” with increased EQ setting. The quality defined by Q corresponds to the quality at full accentuation or lowering. CONSTANT Q: The filter has the same Q at all accentuation or lowering settings. The resulting accentuation or lowering frequency response is not symmetrical.
	4.3	3.0 to 10.0	Q sets the quality of all EQ bands. A high Q value results in a narrowband filter. A low Q value results in a wideband filter.
			The fixed frequencies of EQ bands
			Sets level amplification (accentuation) or reduction (lowering) of a band. A band’s fader is indicated in red when the band has been deactivated by marking the checkbox OFF. Pres- sing the spacebar resets the currently selected fader to 0 dB.
			Deactivating every single EQ-Band is possible by setting this checkbox. Deactivating a band does not change the band’s previously made settings.
			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via “Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (Checkbox OFF Off not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s amplification by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-value of the filters by moving the mouse up or down. For an improved overview the fader of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot.

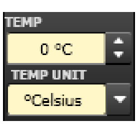
INPUT DELAY
Individual input delays can be set for each input channel of the Dx46.
HINT: The Input Delay parameter is especially useful for delay lines. In this case the required delay depends only on the position of the delay line and is identical for all ways, e.g. output channels of the Dx46. By editing the Input Delay parameter the delays of all output channels routed to this input are adjusted automatically.
You can select the input delay window by clicking onto the fourth block (DLY) in the Flow Diagram Selector or onto the INPUT PROCESSING DELAY block in the flow diagram.



Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding input to any other delay within the same project.
	0 ms	0 to 1000 ms	DELAY allows delaying the corresponding input channel’s audio signal by an adjustable period of time. Entering a value only or a value and unit is possible.
			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the input delay.

General Parameters

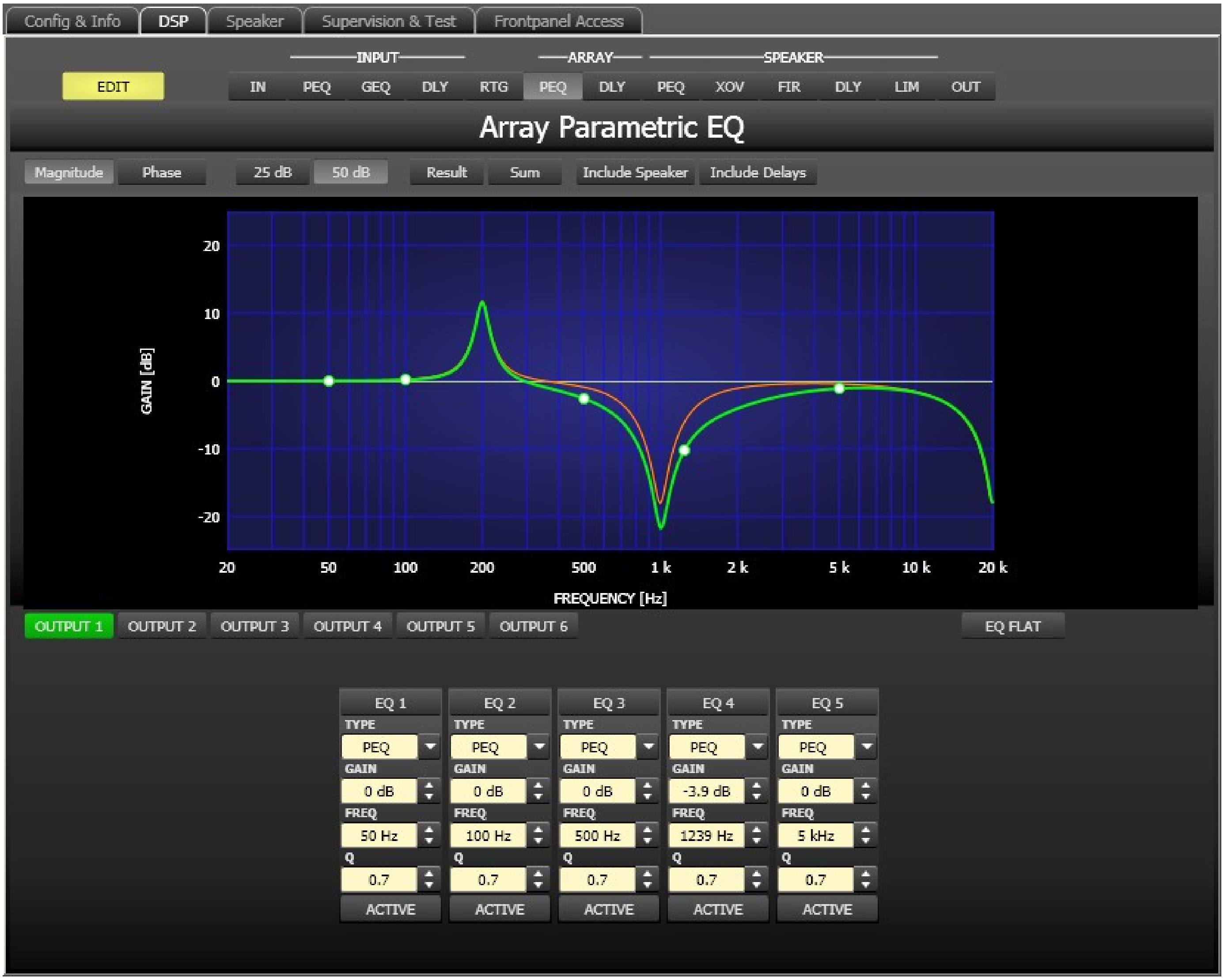
Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

ARRAY PARAMETRIC EQ

All output channels employ 5-band parametric equalizers each, mainly for speaker equalization of arrays. Except for the possibility to select “All pass” as filter type, these filters are identical to the ones of the input EQ's.
The Array Parametric EQ is selected by clicking on the sixth block (PEQ) of the flow diagram selector or onto the ARRAY CONTROL PEQ block in the full-scale flow diagram.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Array Parametric EQ (input PEQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
	Switch for displaying frequency response (magnitude) or phase response (phase)
	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
	Displays the resulting transfer function of all filter and level settings and therefore graphically displaying the audible result at the sound system processor outputs.
	The ”Sum” switch causes display of the summed signal of the output channels, including Output level and Mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally displaying measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other Dx46 Array EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	50 / 100 / 500 / 1k / 5k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..

	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with Allpass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the fil- ter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

Filter Editing via ”Mouse Movement” in the Graphics Display
A white dot in the frequency response display represents an active filter (ACTIVE not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

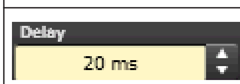
ARRAY DELAY

Individual array delays can be set for each output channel of the Dx46.

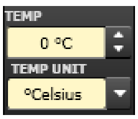
HINT: The Array Delay parameter can be used for adjusting the individual cabinets within a loudspeaker cluster, such as a subwoofer array or a center loudspeaker cluster. For example, in a speaker cluster consisting of two horn-loaded loudspeakers, it is helpful to apply 3-5 ms of delay to one of the loudspeakers in the cluster to improve the coverage in the overlap of the horn patterns. Additionally, the array delay provides a convenient section to apply dedicated delay to individual subwoofer cabinets to create gradient or beam-formed arrays.

You can select the Array Delay window by clicking onto the seventh block (DLY) in the Flow Diagram Selector or onto the ARRAY CONTROL DELAY block in the flow diagram.

Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
	0 ms	0 to 1000 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
			The caption of this button indicates the current state of the delay. Press the ACTIVE button to deactivate the delay.

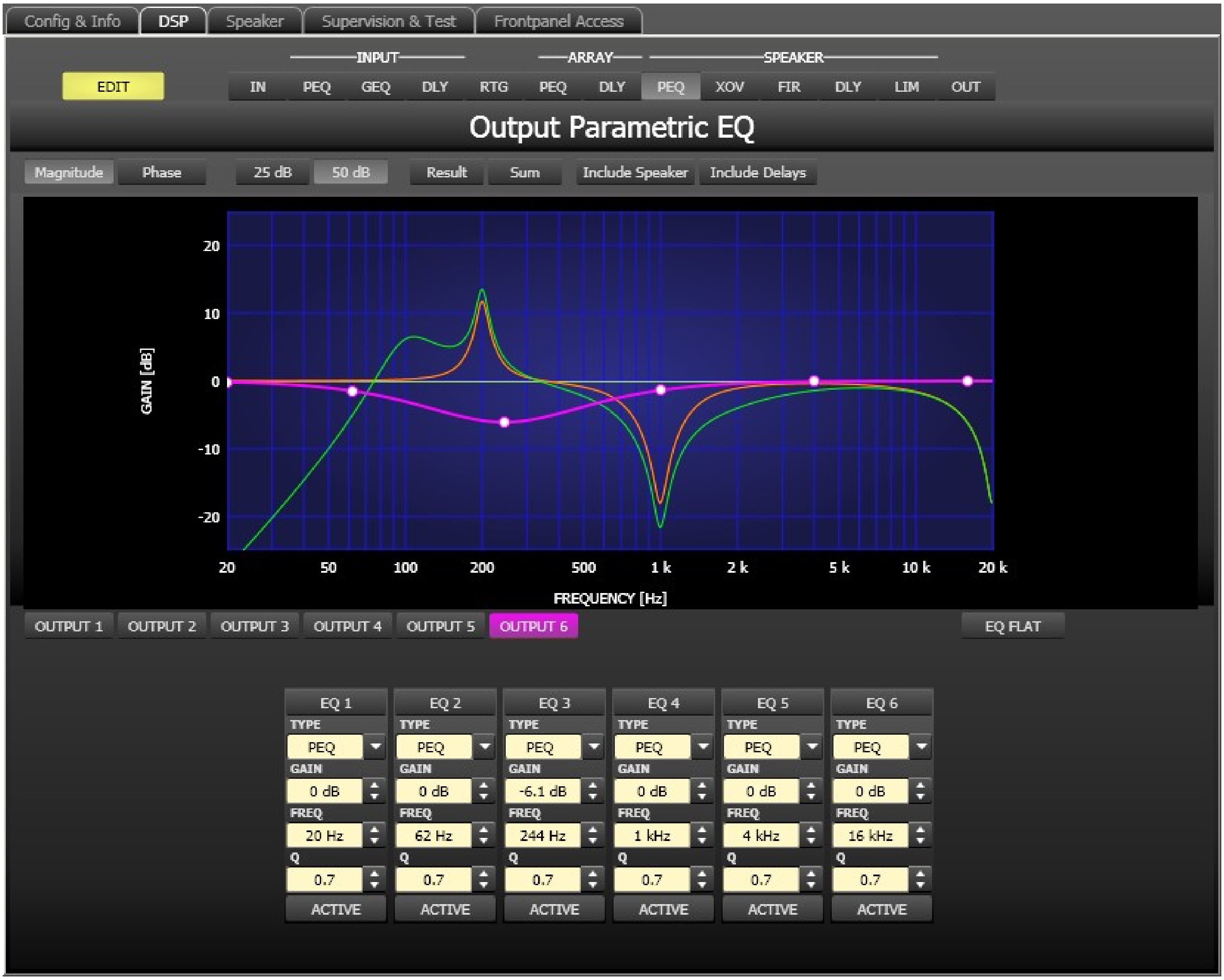
General Parameters

Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display
The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT PARAMETRIC EQ

All output channels employ 6-band parametric equalizers each, mainly for speaker equalization. Except for the possibility to select ”All pass” as filter type, these filters are identical to the ones of the input EQ's.
The Output Parametric EQ is selected by clicking on the eight block (PEQ) of the flow diagram selector or on the SPEAKER PROCESSING PEQ block in the full-scale flow diagram.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Display generally includes all effects of filters that are located pre Output Parametric EQ, which always provides precise overview and control of the resulting frequency response at this point.

Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs.
Sum	The ”Sum” switch causes display of the summed signal of the output channels, including output level and mute. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.

Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels' summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all EQs of the corresponding output to any other Dx46 Output EQ-filter bank within the same project.

Filter Parameters

Element	Default	Range	Description
EQ 1			Name of the corresponding filter band. A click with the right mouse button on this field opens the ”Copy & Paste” menu, which allows convenient copying all EQ-parameters of the according filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. PEQ is a parametric Peak-Dip-Filter with programmable frequency, Q and gain. Loshelv / Hishelv creates a low shelving respectively high shelving equalizer with the following editable parameters: frequency, slope and gain. Lopass / Hipass creates low pass respectively high pass filters with adjustable frequency and slope. Allpass is a filter which only affects the phase but not the frequency response of the transmission function.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/ Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible.
FREQ 31 Hz	20 / 62 / 250 / 1k / 4k / 16k Hz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.

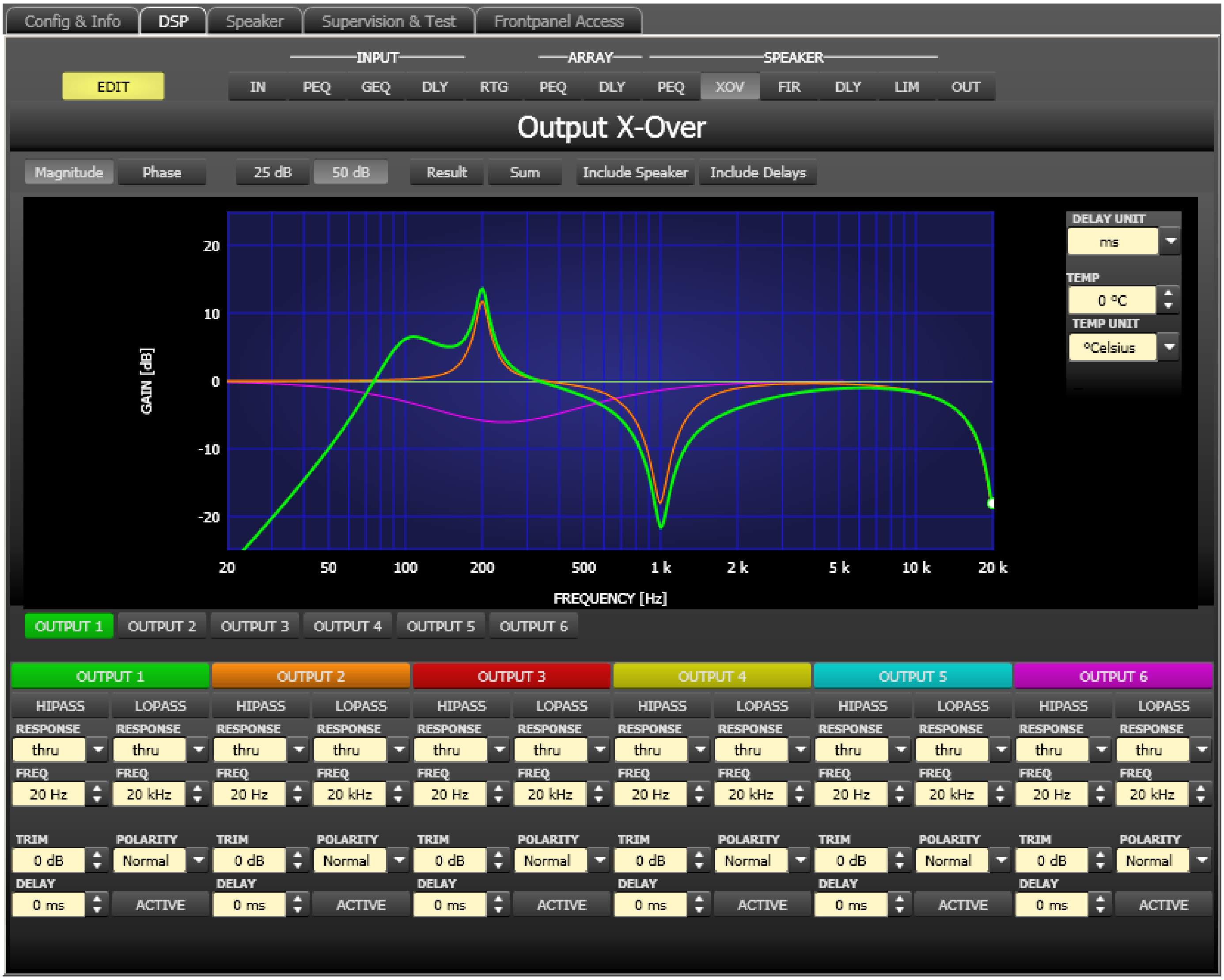
	0.7	0.4 to 40.0 (PEQ), 0.4 to 2.0 (Hi-/Lo-/ Allpass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/oct..
	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
	first	first, second	ORDER (only available with Allpass filters) sets the desired filter order of an All pass filter. A 1st order All pass filter rotates the phase by 180°, a 2nd order All pass filter rotates the phase by 360°.
			The caption of this button indicates the current state of the filter. Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.
			Press the EQ FLAT button to reset the gain of all filters to 0 dB.

Filter Editing via ”Mouse Movement” in the Graphics Display

A white dot in the frequency response display represents an active filter (BYPASS not engaged). Clicking with the left mouse button on this dot and keeping the mouse button pressed down allows changing the selected filter’s frequency by moving the mouse to the left or to the right as well as its gain or cut (depending on the selected filter type) by moving the mouse up or down. Clicking with the right mouse button on the white dot and keeping the mouse button pressed down allows changing the Q-values. For an improved overview the name of the corresponding filter band appears in color as soon as the mouse cursor is positioned over its white dot. An additional white graph indicates the frequency response of the actually selected filter.

OUTPUT X-OVER

The Output X-Over window allows accessing the frequency crossover with Hi- and Lo-Pass filters, a delay, gain-trim and polarity selector switch. By means of these parameters you are able to correctly configure a multi-way speaker system’s individual frequency bands, compensate for natural delays and adjust levels.
Clicking on the ninth block (XOV)in the Flow Diagram Selector or on the SPEAKER PROCESSING X-OVER block in the large signal flow diagram opens the X-Over window.



Graphics Display Indication

The graphics display offers several different display modes, as described in the following table. Indication generally includes all effects of filters that are located pre X-Over (e.g. Array Parametric EQ), which always provides precise overview and control of the resulting frequency response at this point.

Element	**Description**
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs. The audible result is displayed in bright colors while all "electrical" graphs are drawn in dark colors.
Sum	The "Sum" switch causes display of the summed signal of the output channels. If the "Sum " switch is not pressed the output channels' transfer functions are indicated separately.

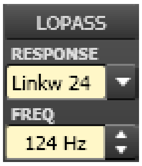
	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

Element	**Description**
	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all X-Over settings of the corresponding output to any other X-Over within the same project.

Channel parameter

Element	**Default**	**Range**	**Description**
	thru, 35 Hz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the HIPASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the HIPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding HI-PASS filter to any HI-PASS filters within the same project.

	thru, 16 kHz	**RESPONSE:** thru, 6dB, 12dB/Q=0.5, 12dB/Q=0.6, 12dB/Q=0.7, 12dB/Q=0.8, 12dB/Q=1.0, 12dB/Q=1.2, 12dB/Q=1.5, 12dB/Q=2.0, Bessel 12dB, Butterworth 12dB, Linkwitz/Riley 12dB, Bessel 18dB, Butterworth 18dB, Bessel 24dB, Butterworth 24dB, Linkwitz/ Riley 24dB **FREQ:** 20 Hz to 20 kHz	This parameter block represents the LOPASS filter. Different types of filters (Bessel, Butterworth, Linkwitz/ Riley) with slopes between 6 dB/Oct. and 24 dB/Oct. can be set as filter response. Selecting filter frequencies between 20 Hz and 20 kHz is possible as well. A click with the right mouse button on the LOPASS field opens the Copy & Paste menu, which allows copying all parameters of the corresponding LO-PASS filter to any LO-PASS filters within the same project.
	0 dB	-30 dB to 6 dB	TRIM allows increasing the level of the corresponding channel by up to 6 dB or lowering it by up to 30 dB to allow level adjustment among individual frequency bands.
	Normal	Normal, Inverted	The POLARITY parameter offers the possibility to invert a channels audio signal, i.e. to rotate its phase by 180°. Inverting the signal may become necessary for some specific crossover settings to eliminate the risk of sound cancellation at the crossover frequency. The effect of the polarity parameter becomes obvious when displaying the summed signal of the two amplifier channels (switch set to “Sum”).
	0 ms	0.0 to 20 ms	DELAY allows the delay of the audio signal from the corresponding output by an adjustable period of time. **HINT: The X-Over Delay parameter is used for the alignment of transducers within cabinets. Optimized delay values are included in Electro-Voice Speaker Settings and should not be edited.**
			Press the ACTIVE button to deactivate the delay (Bypass), which allows for quick A / B-evaluation of the actual effect that the delay has on the sound.

Editing X-Over Filters by ”Dragging the Mouse” in the Graphics Display

Active X-Over filters (Response not set to thru) are indicated by a white dot on the frequency response curve, which represents the corresponding filter. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the frequency of the corresponding filter by moving the mouse to the left or the right. A filter’s title ”lights” in color as soon as the mouse cursor is positioned on top of the corresponding white dot to provide improved overview and handling.

OUTPUT FIR

Each output of the Dx46 offers a 512 taps FIR filter.
The Output FIR is selected by clicking on the tenth block (FIR) of the flow diagram selector or on the SPEAKER PROCESSING FIR block in the full-scale flow diagram.

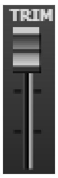
Element	Description
Magnitude Phase	Switch for displaying frequency response (magnitude) or phase response (phase)
25 dB 50 dB	Switch for scaling the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)
Result	Displays the resulting transfer function of all filter and level trim settings and therefore graphically displaying the audible result at the sound system processor outputs. The audible result is displayed in bright colors while all ”electrical” graphs are drawn in dark colors.
Sum	The ”Sum” switch causes display of the summed signal of the output channels. If the ”Sum ” switch is not pressed the output channels’ transfer functions are indicated separately.
Include Delays	Switch for including programmed delays in the frequency or phase response indication. The delays mainly affect phase response indication. Indicating the sound system processor channels’ summed signals reveals very clearly the effect that the delays have on the frequency response, e.g. as notch filter effect.
Include Speaker	Switch for additionally indicating measured speaker transfer functions. For this function to be effective you first have to load speaker data in the ”Speaker” tab.

Channel Selection

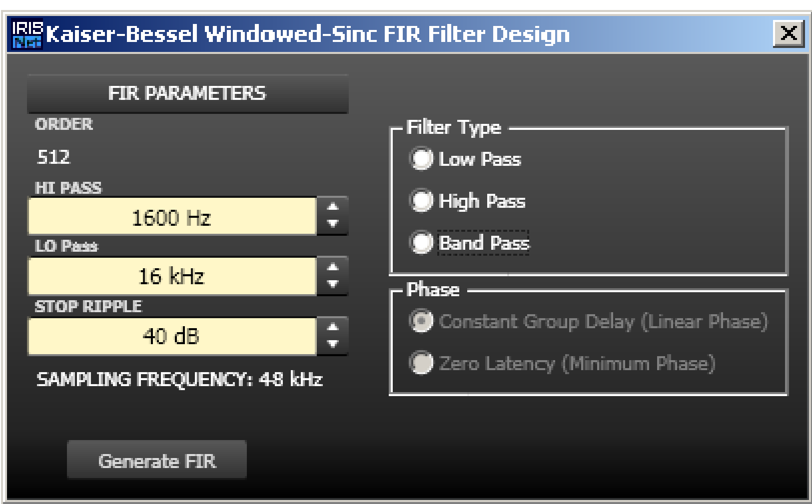
Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for filter editing. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying of all FIR filter settings of the corresponding output to any other FIR filter within the same project.

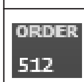
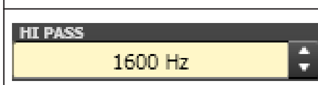
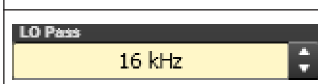
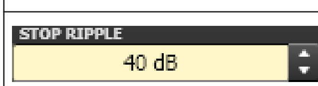
Channel Parameters

Element	Description
FIR INFO IRIS-Net FIR-Filter=Thru	Description of the FIR filter currently in use.
LOAD	After clicking onto LOAD the "Open File..." dialog box appears. Enter the correct path of the directory in which the desired file is located and select the desired FIR file to be opened. This loads and afterwards displays all FIR filter parameters that are stored within that file. **CAUTION: The loaded FIR filter file becomes instantly audible when in on-line mode. Be sure to select the desired FIR file with the correct set of parameters. In the worst case, this could lead to severe damage to the connected loudspeaker cabinets due to improper signal processing!**
EXPORT	After clicking on EXPORT FIR a "Save File..." dialog box appears. Enter the correct path of the directory that you want to save the data in. Enter a file name (without extension). Click on the SAVE button to store the FIR filter parameters together with the corresponding file name. ".gkf" is automatically added as file extension.
CLEAR	Clears the current FIR filter settings. A Default-FIR-Filter (Thru) is activated instead.
NEW	Clicking on the NEW button opens the Filter Design dialog.

	Press the ACTIVE button to deactivate the filter (Bypass), which allows for quick A / B-evaluation of the actual effect that the filter has on the sound.
	Adjusts the gain of the signal between -30 dB and +6 dB.
	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
	Clicking the MUTE button with the left mouse button mutes the corresponding output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the output is again active. The MUTE button is virtually disengaged and not lit.

FIR-Filter Design



Element	**Default**	**Range**	**Description**
			ORDER of the FIR filter.
	1600 Hz	20 to 19999 Hz	HI PASS sets the cut-off frequency of the Hi pass filter.
	16 kHz	21 to 20000 Hz	LO Pass sets the cut-off frequency of the Lo pass filter.
	40 dB	21 to 100 dB	STOP RIPPLE sets the slope of the FIR filter.

			Allows selection of the FIR filter type of the corresponding output channel.
			Press this button to generate the FIR filter.

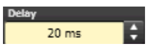
OUTPUT DELAY

Individual output delays can be set for each output channel of the Dx46.

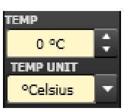
HINT: The Dx46's output delays can be used to compensate for the positioning of cabinets or speaker arrays relative to each other or the original sound source, for example aligning the PA to the stage or aligning the full-range loudspeakers to the subwoofers. The Output Delay parameter determines the delay time of the corresponding channel or the distance between different loudspeaker clusters.

You can select the output delay window by clicking onto the eleventh block (DLY) in the Flow Diagram Selector or on the SPEAKER PROCESSING DELAY block in the flow diagram.

Channel Parameters

Element	Default	Range	Description
			Channel name. A click with the right mouse button opens the ”Copy & Paste” menu, which allows convenient copying all delay parameters of the corresponding output to any other delay within the same project.
	0 ms	0 to 1000 ms	Delay allows delaying the corresponding output channel’s audio signal by an adjustable period of time.
			Press the ACTIVE button to deactivate the output delay.

General Parameters

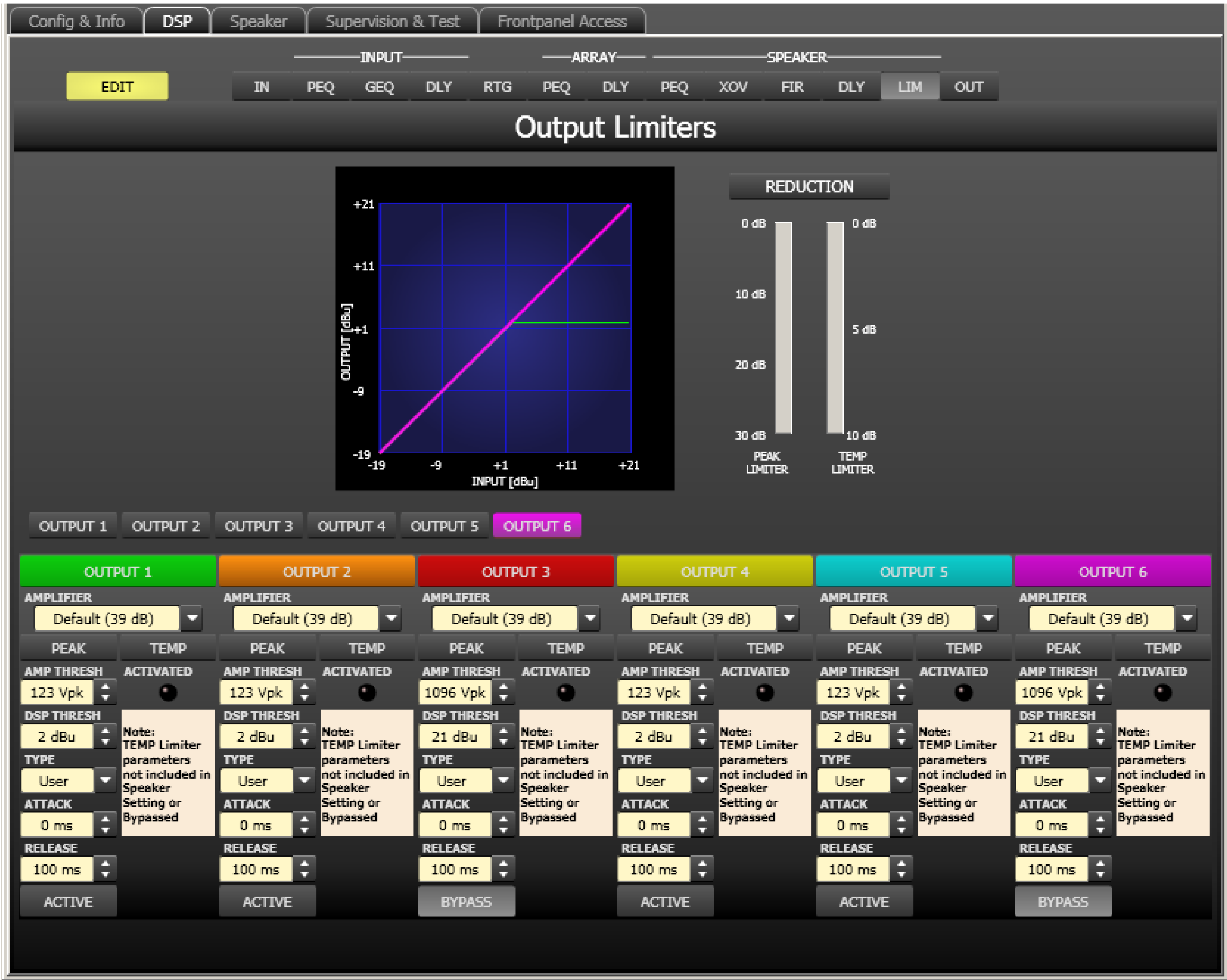
Element	Default	Range	Description
	ms	ms, samples, ft, in, mtr, cm, µs, s	This lets you select the unit of measurement for the delays.
	0 °Celsius	-20 to 60 °C -4 to 140 °F	Entering the actual ambient temperature is possible here. In case you have chosen a distance value as unit of measurement for the delay, delay times are corrected in relation to temperature. Temperatures can be entered as °C or °F.

Editing Delays by ”Dragging the Mouse” in the Graphics Display

The graphics display shows the corresponding speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected channel’s delay time. A delay’s title is shown black as soon as the mouse cursor is positioned on top of the corresponding icon to provide improved overview and handling.

OUTPUT LIMITERS

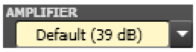
Each output channel of the sound system processor offers a peak limiter and a TEMP limiter. These functions can be accessed via the Output Limiters window to change the corresponding parameters providing reliable protection for the connected speaker systems against sudden peaks and overload.
Clicking on the 12. block (LIM) in the Flow Diagram Selector or double clicking on the SPEAKER PROCESSING LIMITERS block in the large flow diagram opens the Output Limiters window.



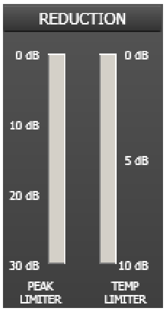
Channel Selection

Element	Description
OUTPUT 1	Switch for selecting output 1, 2, 3, 4. 5 or output 6 for limiter editing. A click with the right mouse button opens the "Copy & Paste" menu, which allows convenient copying all limiter settings of the corresponding output to any other limiter within the same project.

Limiter Parameters

Element	Default	Range	Description
	Default (39 dB)	User, Default (39 dB), Q44, Q66, CP1200, CP1800, CP2200, CP3000S, CP4000S, P1200 (0dBu), P1200 (+6dBu), P1200 (26dB), P2000 (0dBu), P2000 (+6dBu), P2000 (26dB), P3000 (0dBu), P3000 (+6dBu), P3000 (26dB), TG5 (0dBu), TG5 (32dB), TG5 (35dB), TG7 (0dBu), TG7 (32dB), TG7 (35dB), Q44 MKII, Q66 MKII, Q99, Q1212, CPS2.4, CPS2.6, CPS2.9, CPS2.12	Select the amplifier type connected to output of the Dx46.
	123 Vpk		AMP THRESH determines the audio signal level above which the peak limiter starts operating.
	2 dBu		DSP THRESH determines the audio signal level above which the peak limiter starts operating. This value may change depending on the AMP- LIFIER type that is selected, as the sensitivity and output power are automatically calculated with the Vpk value to provide DSP Threshold.
	User	User, Hi, Mid, Lo, Sub	TYPE allows a user to select a bandpass type, and the software will enter appropriate default time constants for the bandpass selected. Electro- Voice speaker settings include factory-defined time constants, so this section is only for use when creating DSP settings from scratch.
	0 ms	0 to 50 ms	ATTACK determines how fast the limiter reduces amplification when the threshold is exceeded.
	100 ms	10 bis 1000 ms	RELEASE determines how fast the limiter returns to normal amplification, after the audio signal level declined the threshold.
			Press the ACTIVE button to deactivate the peak limiter.
			The ACTIVATED LED lights green if the TEMP limiter is active.

Gain Reduction Meters

Element	Description
	These indicators show the reduction in dB that is applied to the audio signal by the Peak limiter (PEAK) or the TEMP limiter (TEMP LIMITER). Level reduction is indicated as vertical yellow bar graph.

Editing Limiter Parameters by “Dragging the Mouse” in the Graphics Display
Active limiters (BYPASS button is not engaged) are indicated by a white dot in the graphics display representing its function. A click with the left mouse button onto this dot and keeping the mouse button pressed down lets you set the threshold for the corresponding limiter by vertically dragging the mouse.

Speaker

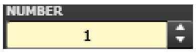
The Speaker Dialog offers the possibility to load the acoustic measurement data of different loudspeaker systems, assign it to the sound system processor channels and display the acoustic results. The speaker system datasets, which are provided as ”speaker files” (*.spk), contain factory-measured frequency- and phase responses of loudspeaker systems.
The speaker data as well as any settings made in this window have no direct influence on the transfer function of the sound system processor. Nevertheless, they provide the user with the possibility for creating loudspeaker systems presets of a higher quality. Overlaying the measured frequency- and phase responses in the equalizer and crossover windows enables the user to customize the filter parameters. The summing display mode shows the result of sound system processor plus speaker transfer functions.
Clicking on the Speaker tab in the Configuration Panel opens the Speaker Dialog.

Indication on the Graphic Display

Element	Description
	Switch for toggling between frequency response (magnitude) and phase response (phase) display
	Switch for adjusting the scale of the dB-axis to 25 dB (± 12.5 dB) or to 50 dB (± 25 dB)

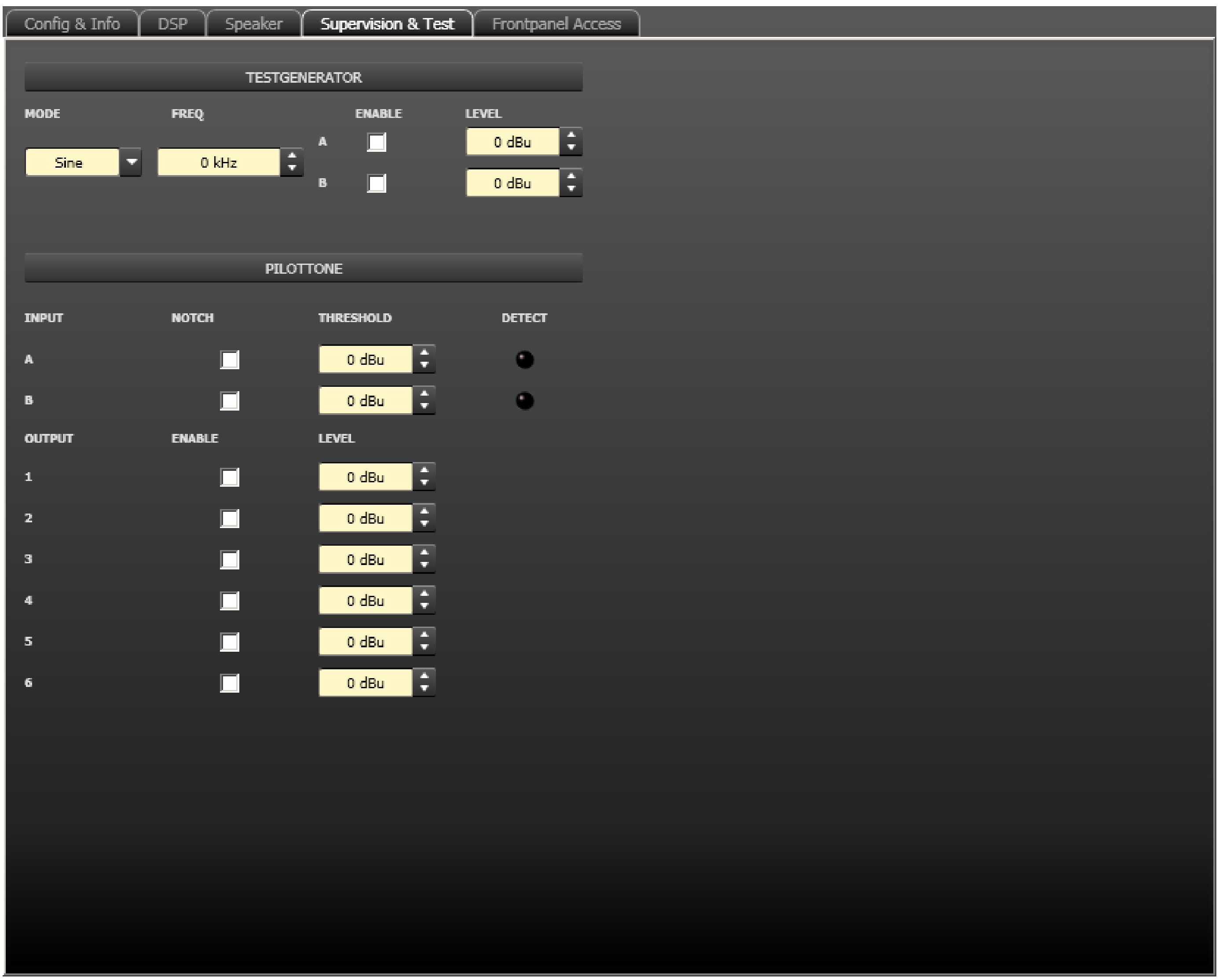
Channel Parameters

Element	Default	Range	Description
			Switch for selecting output 1, 2, 3, 4. 5 or output 6 for limiter editing.
			The name of the loaded loudspeaker model is shown in the black-shaded field.

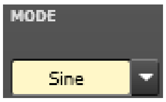
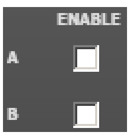
	1	1 to 10	The NUMBER parameter allows the user to specify the number of speaker systems connected to the corresponding channel. Doubling the number of speakers results in a level increase of 6 dB within the selected channel.
			Clicking the LOAD button opens a dialog that allows the selection of the desired speaker file.
			Clicking the CLEAR button clears the previously loaded measured speaker data of the selected channel.

Supervision & Test

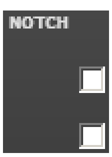
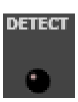
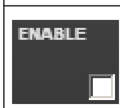
The Supervision window allows configuration of the test and pilot tone generator, additionally the status of the pilot tone detection is indicated.



Test generator

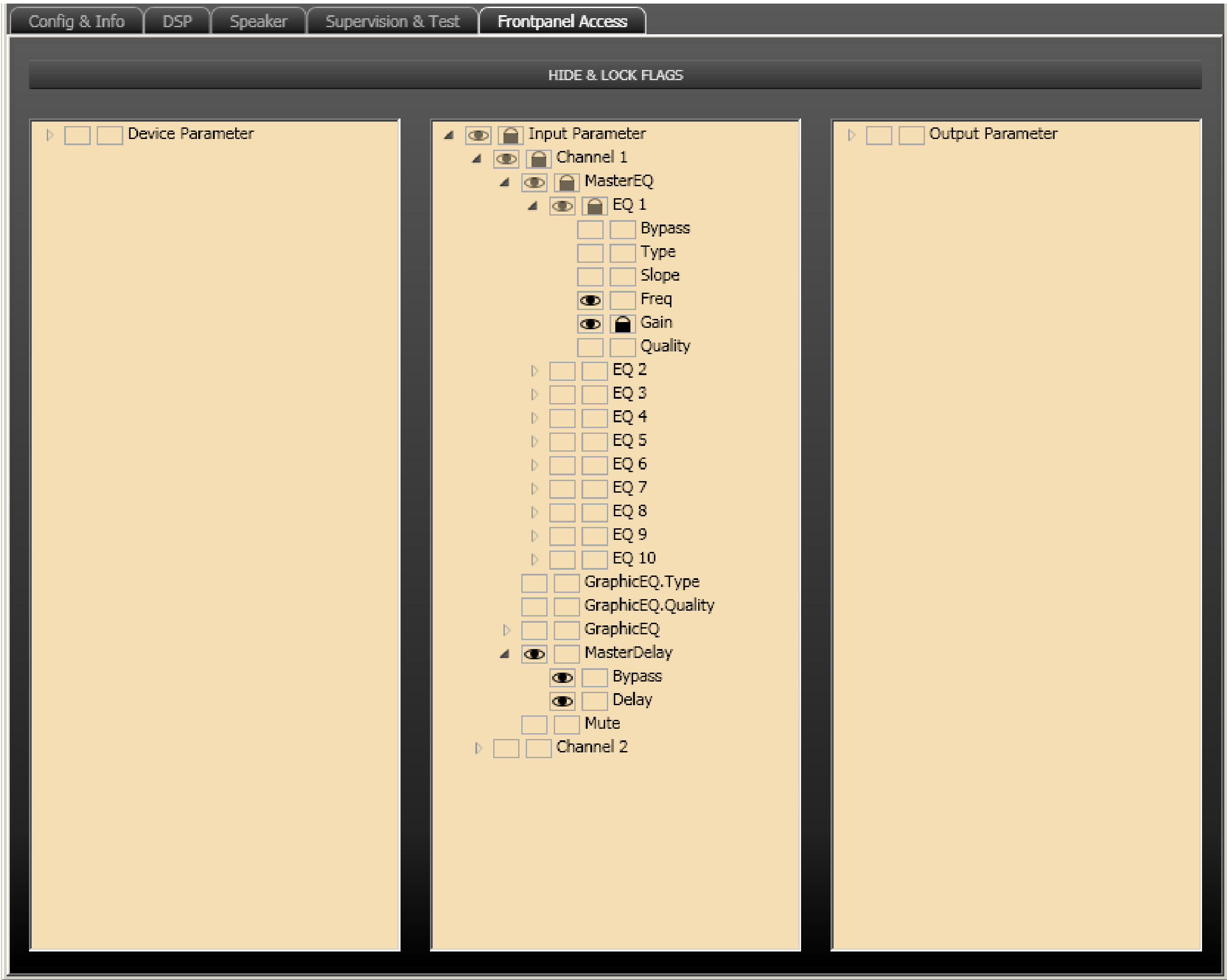
Element	Default	Range	Description
	Sine	Sine, White Noise, Pink Noise	Select the mode of the test generator.
	0 kHz	20 Hz to 20 kHz	Set the frequency of the generated sine signal. The FREQ parameter is only available is MODE Sine is selected.
	Off	On / Off	Activate the test generator for input channel A and/or input channel B.
	0 dBu	-80 to 0 dBu	Enter the signal level for input channel A or B in dBu.

Pilot tone

Element	Default	Range	Description
	Off	On / Off	The checkbox activates a notch filter in input A or B. The notch filter filters an existing pilot tone out of the input signal.
	0 dBu	-80 to 0 dBu	This field sets the pilot tone detection’s threshold.
			The pilot tone detection results in OK (LED lights green) when the level of the pilot signal exceeds the threshold. Without a pilot tone being present or if the signal level is below the set threshold, analysis results in a fault message on the corresponding input channel (LED does not light).
	Off	On / Off	Checkbox for activating or deactivating the pilot tone generator.
	0 dBu	-128 bis 0 dBu	Enter the pilot tone signal level in dBu.

Frontpanel Access

This dialog allows selecting which parameters should be visible and/or editable on the front panel of the Dx46. By default all parameters are visible (eye icon set) and can be edited (lock icon not set). Remove the eye icon of a parameter that should not be visible on the front panel. Activate the lock icon of a parameter that should not be editable at the front panel.


Config & Info
DSP
Speaker
Supervision & Test
Frontpanel Access
HIDE & LOCK FLAGS
Device Parameter
Input Parameter
Channel 1
MasterEQ
EQ 1
Bypass
Type
Slope
Freq
Gain
Quality
EQ 2
EQ 3
EQ 4
EQ 5
EQ 6
EQ 7
EQ 8
EQ 9
EQ 10
GraphicEQ.Type
GraphicEQ.Quality
GraphicEQ
MasterDelay
Bypass
Delay
Mute
Channel 2
Output Parameter

Dx46 and DSP 600

The following chapters are valid for Dx46 and DSP 600.

ASCII Control Protocol

The ASCII Control Protocol allows controlling and polling all parameters. It can be used as interface for the connection of media control systems or facility management systems. Communication is performed via USB port or Ethernet port (telnet) using an easy to implement ASCII protocol which allows easy integration of the device in media and/or touch panel applications. For programming notes and a complete description of the protocol please refer to the following chapters.

RS-232 SETTINGS

When connecting the Dx46/DSP 600 to the PC via USB, the RS-232 interface of the device appears as a virtual port in the operating system. The virtual port is preconfigured for full duplex operation. Set values are:

Parameter	**Setting**
Baud Rate	19200 bits per second
Data Bits	8
Parity	None
Stop Bits	1
Flow Control	Xon / Xoff

The command string "*** Dx46 command mode entered ***" or "*** DSP 600 command mode entered ***" is sent to RS-232 once the device is powered up and has completed its boot sequence. The RS-232 interface is now ready for communication.

ETHERNET SETTINGS

Factory set values of the Ethernet port are:

Parameter	**Setting**
IP address	192.168.1.100
Network mask	255.255.255.0
Standard gateway	192.168.1.1
Telnet port	21

A Telnet session can be established with an application such as Windows HyperTerminal. The command string "*** Dx46 command mode entered ***" or "*** DSP 600 command mode entered ***" will be displayed when the ASCII Control Protocol has started successfully. The device is now ready for ASCII Control Protocol communication.

ASCII CONTROL PROTOCOL

A simple ASCII string protocol, which is referred to as ASCII Control Protocol is implemented in the device. Commands are organized in a tree structure with up to 4 levels. The slash "/" or a space " " can be used for separation. The question mark "?" can be utilized to query parameter settings or commands of the corresponding level. To step down one level you have to enter "..". Use "/" to get back to level 1.

The following table lists the ASCII Control Protocol commands with brief explanations.

Ebene1	**Ebene2**	**Ebene3**	**Ebene4**	**Read Write**	**Values**	**Description**
						Commands for RS232 communication
/COMM	/LINEFEED			R / W	ON, OFF	Linefeed Status for RS232 communication
	/PROMPT			R / W	ON, OFF	Prompt Status for RS232 communication
	/ECHO			R / W	ON, OFF	Echo Status for RS232 communication
						Commands for device front
/FRONT	/CONTRAST			R / W	-10 to +10	Contrast of displays
	/BRIGHTNESS			R / W	0 to 10	Brightness of displays
	/INTENSITY			R / W	0 to 10	Brightness of LEDs
	/VU	/MODE		R / W	FAST, SLOW, PEAK	Mode of VU meter
		/REF		R / W	CLIP, LIMIT	Reference for VU meter
	/SAVER			R / W	ON, OFF	State of screen saver
						Commands for Control Port
/CONTROL	/PINS			R	0, 1	State of control inputs (1 = open, 0 = closed)
	/STATE			R		Indicates the highest number of all control inputs with state „0“.
	/PORT1			R / W		Preset to be loaded when Port 1 is closed.
	...			R / W		
	/PORT5			R / W		Preset to be loaded when Port 5 is closed.

						Commands for Master/Slave settings
/DCMS	/MODE			R / W	OFF, MAS-TER, SLAVE	The mode setting only works if more than one Dx46/DSP 600 device is connected to an Ethernet. Devices that are Master or Slave have always identical parameter settings. Select „MASTER“, if this Dx46/ DSP 600 should write the parameter settings to one or more other Dx46/DSP 600 (Slave). Select „SLAVE“, if the parameter settings of this Dx46/DSP 600 should be read from another Dx46/DSP 600 (master). Select „OFF“, if the parameter settings of this Dx46/DSP 600 should be independent from other devices.
	/ID			R / W	0 to 255	Each master Dx46/DSP 600 connected to the Ethernet must have an unique network id. Enter the id of the Master Dx46/DSP 600 the parameters should be read from if this Dx46/DSP 600 is used as “SLAVE”.
	/MASK			R / W	DEV, A, B, 1, 2, 3, 4, 5, 6	If the mode „SLAVE“ is selected, choose the parameter groups that this DSP 600 should read from the Master DSP 600. Following groups are available: DEV: Parameters of the device A or B: Parameters of input A or B 1 to 6: Parameter of output 1 to 6
						Service commands
/SERVICE	/VER			R		Read software version
	/AES	/LOCKED		R	ON, OFF	Read locked state
		/ERROR		R	ON, OFF	Read error state (OFF = no error)
		/SMPRATE		R		Read sample rate in kHz
	/ETH	/MAC		R / W		Read MAC address
		/IP		R / W		Read or write IP address

		/MASK		R / W		Read or write network mask
		/ GATEWAY		R / W		Read or write default gateway
	/ONTIME			R		Read uptime of the device
	/PILOT			R		Read pilot supervision of inputs
						Commands for DSP parameter and presets
/ PRESET	/PRM	/DEVICE	/IDX00	R / W		Read and write of device parameter values via index numbers. For further details please refer to Device Parameter Index Table
			...			
			/IDX0B	R / W		
		/IN_A	/IDX00	R / W		Read and write of input channel A DSP parameter values via index numbers. For further details please refer to DSP Parameter Index Table.
			...			
			/IDX86	R / W		
		/IN_B	...			Same as above for input channel B
		/OUT_1	/IDX00	R / W		Read and write of output channel 1 DSP parameter values via index numbers. For further details please refer to DSP Parameter Index Table
			...			
			/IDX62	R / W		
		...				
		/OUT_6	...			Same as above for output channel 6
/ PRESET	/LOAD			R / W	U01 to U30, F01 to F60	Read last loaded/saved preset. If the preset was edited, "(edited)" is indicated. Write to load a preset.
	/SAVE			W	U01 to U30	Write to save a user preset.
	/LIST			R		List of all presets.

DEVICE PARAMETER INDEX TABLE

Index	Parameter	Values	Description
/IDX00	lock flag list for device parameters	NONE, 0 to 9	“Set” means locked

/IDX01	hide flag list for device parameters	NONE, 0 to 9	“Set” means hidden
/IDX02	device name	max. 30 characters	
/IDX03	audio input	0 / 1	0 = analog input, 1 = AES input
/IDX04	air temperature	-20 to +60	Air temperature in degree Celsius
/IDX05	Test generator mode	0, 1, 2	0 = sine, 1 = white noise, 2 = pink noise
/IDX06	Test generator frequency	20 to 20000	Hz
/IDX07	sample rate	1	1 = 48 kHz
/IDX08	preset title	max. 30 characters	
/IDX09	preset configuration	0 to 6	0 = 2way stereo+FR 1 = 3way stereo 2 = 4way+FR 3 = 5way+FR 4 = free configuration 5 = 3way stereo monosub 6 = 4way stereo monosub
/IDX0A	analog 6dB input damp	0 / 1	0 = 0 dB 1 = 6dB analog damp at analogue audio inputs
/IDX0B	Edit mode	0 / 1	0 = standard (link handling active) 1 = full (independent parameter access to all channels / xovers)

DSP PARAMETER INDEX TABLE
Input channel A or B

Index	Parameter	Values	Description
/IDX00	lock flag list for device parameters	NONE, 0 to 84	„Set“ means locked
/IDX01	hide flag list for device parameters	NONE, 0 to 84	„Set“ means hidden
/IDX02	Test generator enable	0 / 1	0 = disable, 1 = enable
/IDX03	Test generator level	-80 to 0	signal level in dBu
/IDX04	pilot 19kHz detection threshold	-80 to 0	signal level in dBu
/IDX05	pilot 19kHz notch filter	0 / 1	0 = disable, 1 = enable
/IDX06	parametric eq1 bypass	0 / 1	0 = active, 1 = bypass
/IDX07	parametric eq1 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX08	parametric eq1 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX09	parametric eq1 frequency	20 to 20000	Hz

/IDX0A	parametric eq1 gain	-18 to 12	dB
/IDX0B	parametric eq1 quality	0.4 to 40	
/IDX0C	parametric eq2 bypass	0 / 1	0 = active, 1 = bypass
/IDX0D	parametric eq2 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX0E	parametric eq2 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX0F	parametric eq2 frequency	20 to 20000	Hz
/IDX10	parametric eq2 gain	-18 to 12	dB
/IDX11	parametric eq2 quality	0.4 to 40	
/IDX12	parametric eq3 bypass	0 / 1	0 = active, 1 = bypass
/IDX13	parametric eq3 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX14	parametric eq3 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX15	parametric eq3 frequency	20 to 20000	Hz
/IDX16	parametric eq3 gain	-18 to 12	dB
/IDX17	parametric eq3 quality	0.4 to 40	
/IDX18	parametric eq4 bypass	0 / 1	0 = active, 1 = bypass
/IDX19	parametric eq4 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX1A	parametric eq4 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX1B	parametric eq4 frequency	20 to 20000	Hz
/IDX1C	parametric eq4 gain	-18 to 12	dB
/IDX1D	parametric eq4 quality	0.4 to 40	
/IDX1E	parametric eq5 bypass	0 / 1	0 = active, 1 = bypass
/IDX1F	parametric eq5 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX20	parametric eq5 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX21	parametric eq5 frequency	20 to 20000	Hz
/IDX22	parametric eq5 gain	-18 to 12	dB
/IDX23	parametric eq5 quality	0.4 to 40	
/IDX24	parametric eq6 bypass	0 / 1	0 = active, 1 = bypass
/IDX25	parametric eq6 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX26	parametric eq6 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX27	parametric eq6 frequency	20 to 20000	Hz
/IDX28	parametric eq6 gain	-18 to 12	dB
/IDX29	parametric eq6 quality	0.4 to 40	
/IDX2A	parametric eq7 bypass	0 / 1	0 = active, 1 = bypass

/IDX2B	parametric eq7 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX2C	parametric eq7 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX2D	parametric eq7 frequency	20 to 20000	Hz
/IDX2E	parametric eq7 gain	-18 to 12	dB
/IDX2F	parametric eq7 quality	0.4 to 40	
/IDX30	parametric eq8 bypass	0 / 1	0 = active, 1 = bypass
/IDX31	parametric eq8 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX32	parametric eq8 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX33	parametric eq8 frequency	20 to 20000	Hz
/IDX34	parametric eq8 gain	-18 to 12	dB
/IDX35	parametric eq8 quality	0.4 to 40	
/IDX36	parametric eq9 bypass	0 / 1	0 = active, 1 = bypass
/IDX37	parametric eq9 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX38	parametric eq9 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX39	parametric eq9 frequency	20 to 20000	Hz
/IDX3A	parametric eq9 gain	-18 to 12	dB
/IDX3B	parametric eq9 quality	0.4 to 40	
/IDX3C	parametric eq10 bypass	0 / 1	0 = active, 1 = bypass
/IDX3D	parametric eq10 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX3E	parametric eq10 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX3F	parametric eq10 frequency	20 to 20000	Hz
/IDX40	parametric eq10 gain	-18 to 12	dB
/IDX41	parametric eq10 quality	0.4 to 40	
/IDX42	graphic eq type	0 to 2	0 = symQ 1 = const.Q 2 = prop.Q
/IDX43	graphic eq quality	3 to 10	
/IDX44	graphic eq bypass band 20Hz	0 / 1	0 = active, 1 = bypass
/IDX45	graphic eq gain band 20Hz	-12 to +12	dB
/IDX46	graphic eq bypass band 25Hz	0 / 1	0 = active, 1 = bypass
/IDX47	graphic eq gain band 25Hz	-12 to +12	dB

/IDX48	graphic eq bypass band 31.5Hz	0 / 1	0 = active, 1 = bypass
/IDX49	graphic eq gain band 31.5Hz	-12 to +12	dB
/IDX4A	graphic eq bypass band 40Hz	0 / 1	0 = active, 1 = bypass
/IDX4B	graphic eq gain band 40Hz	-12 to +12	dB
/IDX4C	graphic eq bypass band 50Hz	0 / 1	0 = active, 1 = bypass
/IDX4D	graphic eq gain band 50Hz	-12 to +12	dB
/IDX4E	graphic eq bypass band 63Hz	0 / 1	0 = active, 1 = bypass
/IDX4F	graphic eq gain band 63Hz	-12 to +12	dB
/IDX50	graphic eq bypass band 80Hz	0 / 1	0 = active, 1 = bypass
/IDX51	graphic eq gain band 80Hz	-12 to +12	dB
/IDX52	graphic eq bypass band 100Hz	0 / 1	0 = active, 1 = bypass
/IDX55	graphic eq gain band 100Hz	-12 to +12	dB
/IDX54	graphic eq bypass band 125Hz	0 / 1	0 = active, 1 = bypass
/IDX55	graphic eq gain band 125Hz	-12 to +12	dB
/IDX56	graphic eq bypass band 160Hz	0 / 1	0 = active, 1 = bypass
/IDX57	graphic eq gain band 160Hz	-12 to +12	dB
/IDX58	graphic eq bypass band 200Hz	0 / 1	0 = active, 1 = bypass
/IDX59	graphic eq gain band 200Hz	-12 to +12	dB
/IDX5A	graphic eq bypass band 250Hz	0 / 1	0 = active, 1 = bypass
/IDX5B	graphic eq gain band 250Hz	-12 to +12	dB

/IDX5C	graphic eq bypass band 315Hz	0 / 1	0 = active, 1 = bypass
/IDX5D	graphic eq gain band 315Hz	-12 to +12	dB
/IDX5E	graphic eq bypass band 400Hz	0 / 1	0 = active, 1 = bypass
/IDX5F	graphic eq gain band 400Hz	-12 to +12	dB
/IDX60	graphic eq bypass band 500Hz	0 / 1	0 = active, 1 = bypass
/IDX61	graphic eq gain band 500Hz	-12 to +12	dB
/IDX62	graphic eq bypass band 630Hz	0 / 1	0 = active, 1 = bypass
/IDX66	graphic eq gain band 630Hz	-12 to +12	dB
/IDX64	graphic eq bypass band 800Hz	0 / 1	0 = active, 1 = bypass
/IDX65	graphic eq gain band 800Hz	-12 to +12	dB
/IDX66	graphic eq bypass band 1kHz	0 / 1	0 = active, 1 = bypass
/IDX67	graphic eq gain band 1kHz	-12 to +12	dB
/IDX68	graphic eq bypass band 1.25kHz	0 / 1	0 = active, 1 = bypass
/IDX69	graphic eq gain band 1.25kHz	-12 to +12	dB
/IDX6A	graphic eq bypass band 1.6kHz	0 / 1	0 = active, 1 = bypass
/IDX6B	graphic eq gain band 1.6Hz	-12 to +12	dB
/IDX6C	graphic eq bypass band 2kHz	0 / 1	0 = active, 1 = bypass
/IDX6D	graphic eq gain band 2kHz	-12 to +12	dB
/IDX6E	graphic eq bypass band 2.5kHz	0 / 1	0 = active, 1 = bypass
/IDX6F	graphic eq gain band 2.5kHz	-12 to +12	dB

/IDX70	graphic eq bypass band 3.15kHz	0 / 1	0 = active, 1 = bypass
/IDX71	graphic eq gain band 3.15kHz	-12 to +12	dB
/IDX72	graphic eq bypass band 4kHz	0 / 1	0 = active, 1 = bypass
/IDX73	graphic eq gain band 4kHz	-12 to +12	dB
/IDX74	graphic eq bypass band 5kHz	0 / 1	0 = active, 1 = bypass
/IDX75	graphic eq gain band 5kHz	-12 to +12	dB
/IDX76	graphic eq bypass band 6.3kHz	0 / 1	0 = active, 1 = bypass
/IDX77	graphic eq gain band 6.3kHz	-12 to +12	dB
/IDX78	graphic eq bypass band 8kHz	0 / 1	0 = active, 1 = bypass
/IDX79	graphic eq gain band 8kHz	-12 to +12	dB
/IDX7A	graphic eq bypass band 10kHz	0 / 1	0 = active, 1 = bypass
/IDX7B	graphic eq gain band 10kHz	-12 to +12	dB
/IDX7C	graphic eq bypass band 12.5kHz	0 / 1	0 = active, 1 = bypass
/IDX7D	graphic eq gain band 12.5kHz	-12 to +12	dB
/IDX7E	graphic eq bypass band 16kHz	0 / 1	0 = active, 1 = bypass
/IDX7F	graphic eq gain band 16kHz	-12 to +12	dB
/IDX80	graphic eq bypass band 20kHz	0 / 1	0 = active, 1 = bypass
/IDX81	graphic eq gain band 20kHz	-12 to +12	dB
/IDX82	delay bypass	0 / 1	0 = active, 1 = bypass
/IDX83	delay	0 to 1000	default milliseconds, units can be appended (ms, samples, feet, inch, meter, cm, us, sec)
/IDX84	input mute	0 / 1	0 = normal, 1 = muted

/IDX85	input channel name	max. 30 characters	
/IDX86	audio input gain	-60 to +12	audio input gain in dB

Output channel 1 to 6

Index	Parameter	Values	Description
/IDX00	lock flag list for output parameters	NONE, 0 to 61	„Set“ means locked
/IDX01	hide flag list for output parameters	NONE, 0 to 61	„Set“ means hidden
/IDX02	connected amp model	ASCII chars	Default, User or Dynacord/ EV amp types
/IDX03	gain of amp model „User“	0 to 60	dB
/IDX04	Pilot generator enable	0 / 1	0 = off, 1 = on
/IDX05	Pilot generator level	-128 to 0	dB
/IDX06	route	A, A+B, B	selects the input signal
/IDX07	array delay bypass	0 / 1	0 = active, 1 = bypass
/IDX08	array delay	0 to 100	default milliseconds, units can be appended (ms, samples, feet, inch, meter, cm, us, sec)
/IDX09	xover delay bypass	0 / 1	0 = active, 1 = bypass
/IDX0A	xover delay	0 to 20	default milliseconds, units can be appended (ms, samples, feet, inch, meter, cm, us, sec)
/IDX0B	output delay bypass	0 / 1	0 = active, 1 = bypass
/IDX0C	output delay	0 to 1000	default milliseconds, units can be appended (ms, samples, feet, inch, meter, cm, us, sec)
/IDX0D	array peq 1 bypass	0 / 1	0 = active, 1 = bypass
/IDX0E	array peq 1 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX0F	array peq 1 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX10	array peq 1 frequency	20 to 20000	Hz
/IDX11	array peq 1 gain	-18 to +12	dB
/IDX12	array peq 1 quality	0.4 to 40	
/IDX13	array peq 2 bypass	0 / 1	0 = active, 1 = bypass
/IDX14	array peq 2 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX15	array peq 2 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX16	array peq 2 frequency	20 to 20000	Hz

/IDX17	array peq 2 gain	-18 to +12	dB
/IDX18	array peq 2 quality	0.4 to 40	
/IDX19	array peq 3 bypass	0 / 1	0 = active, 1 = bypass
/IDX1A	array peq 3 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX1B	array peq 3 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX1C	array peq 3 frequency	20 to 20000	Hz
/IDX1D	array peq 3 gain	-18 to +12	dB
/IDX1E	array peq 3 quality	0.4 to 40	
/IDX1F	array peq 4 bypass	0 / 1	0 = active, 1 = bypass
/IDX20	array peq 4 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX21	array peq 4 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX22	array peq 4 frequency	20 to 20000	Hz
/IDX23	array peq 4 gain	-18 to +12	dB
/IDX24	array peq 4 quality	0.4 to 40	
/IDX25	array peq 5 bypass	0 / 1	0 = active, 1 = bypass
/IDX26	array peq 5 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX27	array peq 5 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX28	array peq 5 frequency	20 to 20000	Hz
/IDX29	array peq 5 gain	-18 to +12	dB
/IDX2A	array peq 5 quality	0.4 to 40	
/IDX2B	speaker peq 1 bypass	0 / 1	0 = active, 1 = bypass
/IDX2C	speaker peq 1 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX2D	speaker peq 1 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX2E	speaker peq 1 frequency	20 to 20000	Hz
/IDX2F	speaker peq 1 gain	-18 to +12	dB
/IDX30	speaker peq 1 quality	0.4 to 40	
/IDX31	speaker peq 2 bypass	0 / 1	0 = active, 1 = bypass
/IDX32	speaker peq 2 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX33	speaker peq 2 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX34	speaker peq 2 frequency	20 to 20000	Hz

/IDX35	speaker peq 2 gain	-18 to +12	dB
/IDX36	speaker peq 2 quality	0.4 to 40	
/IDX37	speaker peq 3 bypass	0 / 1	0 = active, 1 = bypass
/IDX38	speaker peq 3 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX39	speaker peq 3 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX3A	speaker peq 3 frequency	20 to 20000	Hz
/IDX3B	speaker peq 3 gain	-18 to +12	dB
/IDX3C	speaker peq 3 quality	0.4 to 40	
/IDX3D	speaker peq 4 bypass	0 / 1	0 = active, 1 = bypass
/IDX3E	speaker peq 4 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX3F	speaker peq 4 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX40	speaker peq 4 frequency	20 to 20000	Hz
/IDX41	speaker peq 4 gain	-18 to +12	dB
/IDX42	speaker peq 4 quality	0.4 to 40	
/IDX43	speaker peq 5 bypass	0 / 1	0 = active, 1 = bypass
/IDX44	speaker peq 5 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX45	speaker peq 5 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX46	speaker peq 5 frequency	20 to 20000	Hz
/IDX47	speaker peq 5 gain	-18 to +12	dB
/IDX48	speaker peq 5 quality	0.4 to 40	
/IDX49	speaker peq 6 bypass	0 / 1	0 = active, 1 = bypass
/IDX4A	speaker peq 6 type	0 to 5	0 = peq, 1 = loshelv, 2 = hishelv, 3 = locut, 4 = hicut, 5 = allpass
/IDX4B	speaker peq 6 slope	1 / 2	1 = 6dB, 2 = 12dB
/IDX4C	speaker peq 6 frequency	20 to 20000	Hz
/IDX4D	speaker peq 6 gain	-18 to +12	dB
/IDX4E	speaker peq 6 quality	0.4 to 40	
/IDX4F	hipass xover type	0 to 17	0 = off, 1 = butter6, 2 = s12q05, 3 = s12q06, 4 = s12q07, 5 = s12q08, 6 = s12q10, 7 = s12q12, 8 = s12q15, 9 = s12q20, 10 = bessel12, 11 = butter12, 12 = linkwz12, 13 = bessel18, 14 = butter18, 15 = bessel24, 16 = butter24, 17 = linkwz24

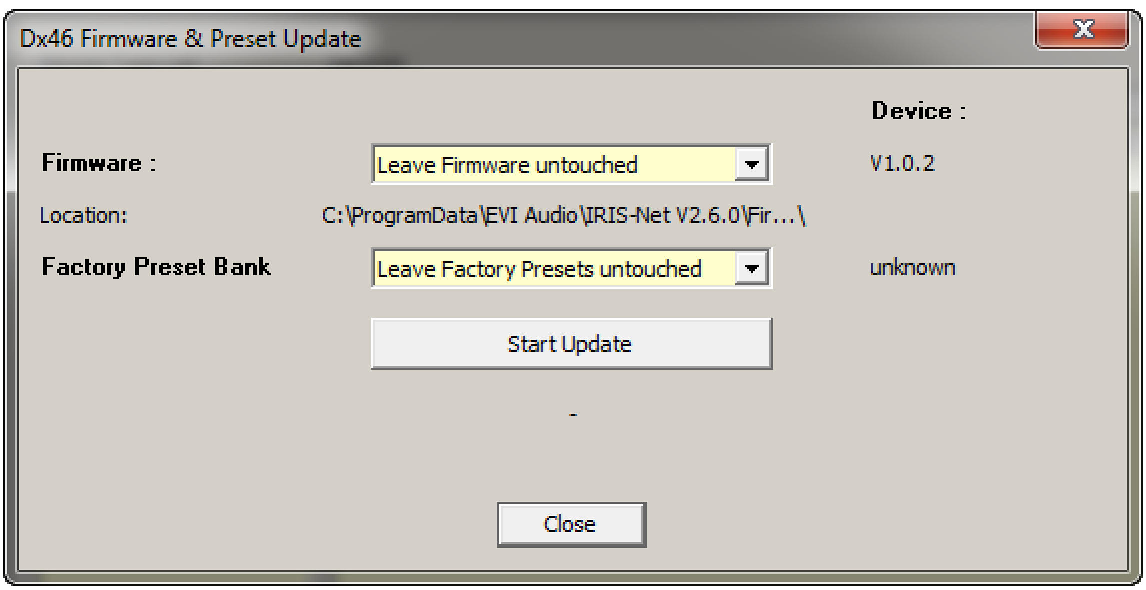
/IDX50	hipass xover frequency	20 to 20000	Hz
/IDX51	lopass xover type	0 to 17	0 = off, 1 = butter6, 2 = s12q05, 3 = s12q06, 4 = s12q07, 5 = s12q08, 6 = s12q10, 7 = s12q12, 8 = s12q15, 9 = s12q20, 10 = bessel12, 11 = butter12, 12 = linkwz12, 13 = bessel18, 14 = butter18, 15 = bessel24, 16 = butter24, 17 = linkwz24
/IDX52	lopass xover frequency	20 to 20000	Hz
/IDX53	FIR filter bypass	0 / 1	0 = active, 1 = bypass
/IDX54	FIR filter	max. 41 characters	read only
/IDX55	polarity	0 / 1	0 = Normal, 1 = Invertiert
/IDX56	mute	0 / 1	0 = ON, 1 = MUTE
/IDX57	level	-128 to +6	dB
/IDX58	level trim	-30 to +6	dB
/IDX59	limiter bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX5A	limiter type	0 to 4	0 = user, 1 = hi, 2 = mid, 3 = lo, 4 = sub
/IDX5B	limiter threshold	+11 to +71	dBu at amp output
/IDX5C	limiter attack time	0 to 50	milliseconds
/IDX5D	limiter release time	10 to 999	milliseconds
/IDX5E	config type	0 to 15	binary LED pattern
/IDX5F	config description	max. 18 characters	
/IDX60	input channel name	max. 30 characters	
/IDX61	thermo limiter bypass	0 / 1	0 = ON, 1 = BYPASS
/IDX62	thermo limiter setting	set of 12 values	whitespace separated complete list of values in the order: r_vc c_vc r_m c_m z brms ctime atime rtime t_max t_wall t_knee

Updating the firmware

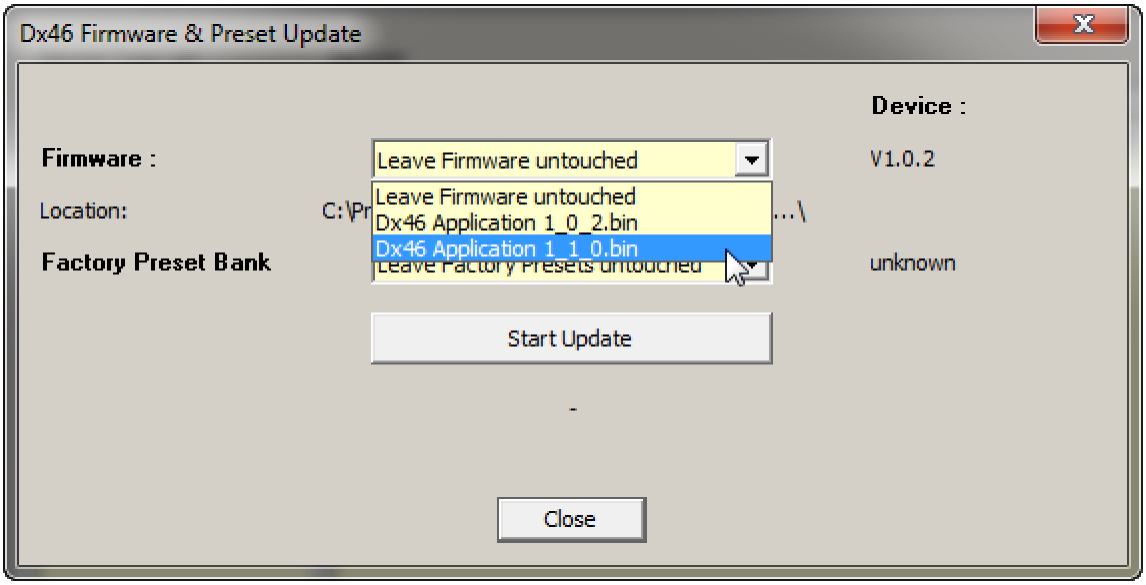
The Device Scan dialog allows updating the firmware of the Dx46/DSP 600.

Caution!
The Dx46/DSP 600 firmware should be updated only, if problems with the firmware used so far exist and these can be fixed by using a new firmware version.
Consequences

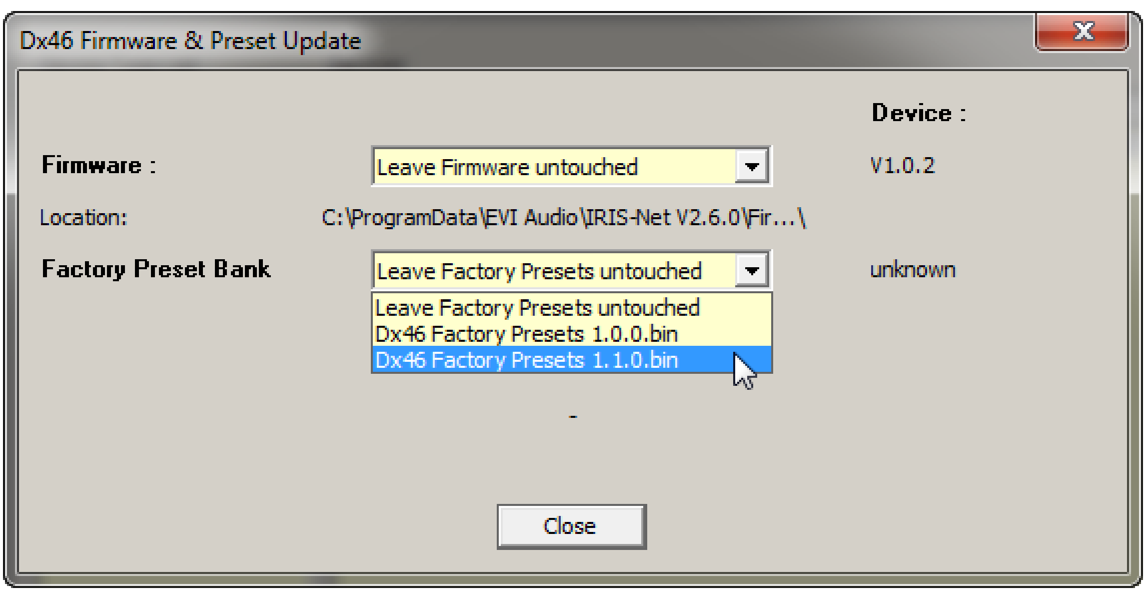
1. Connect the Dx46/DSP 600 to your PC via Ethernet or USB.
2. Start the IRIS-Net application.
3. Selecting the entry Device Scan in the Tools menu lets you access the Device Scan dialog.
4. Select the entry DX46 or DSP600 in the Device Type List. All found devices are listed in the Devices list.
5. Right click the device to be updated in the Devices list. The Firmware Update Dialog appears.



6. The actual firmware file including version number is indicated and can be selected in the line “Firmware”. The IRIS-Net software package always includes the most up-to-date firmware version. The corresponding file is located in the directory: \IRIS-Net\Firmware\DX46 or \IRIS-Net\Firmware\DSP600. This path also appears in the line “Location”. If you want to install a different (preferably newer) firmware version, you have to copy the corresponding file into this directory first.

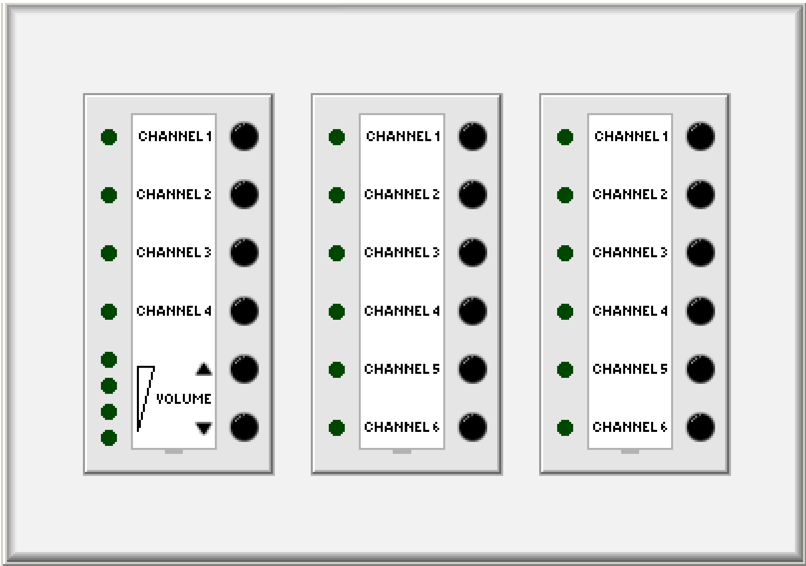


7. The actual firmware file including version number is indicated and can be selected in the line “Factory Preset Bank”. The IRIS-Net software package always includes the most up-to-date factory presets version. The corresponding file is located in the directory: \IRIS-Net\Firmware\DX46 or \IRIS-Net\Firmware\DSP600. This path also appears in the line “Location”. If you want to install a different (preferably newer) factory preset version, you have to copy the corresponding file into this directory first.



8. Clicking onto “Start Update” starts the upgrade procedure. After the upgrade the Dx46/DSP 600 resets and is ready for operation. The upgrade procedure is finished and you can close the dialog window.

PWS PROGRAMMABLE WALL STATION



PWS series wall stations allow local operation and control of different devices and systems via CAN-Bus. Proceeding different functions, like for example source selection, volume control, muting or switching devices on or off is possible. If, in addition, a suitable central control unit is used, controlling building tasks (lighting, heating/air conditioning, etc.) is possible as well.

The minimal configuration for a wall station consists of:

– a front user panel (e.g. PWS-4)
– a CAN-Bus coupler (PWS-C)
– an in IRIS-Net programmed software configuration that has been transferred to the wall station via CAN-Bus
– a to be controlled system controller (e.g. Electro-Voice NetMax N8000)

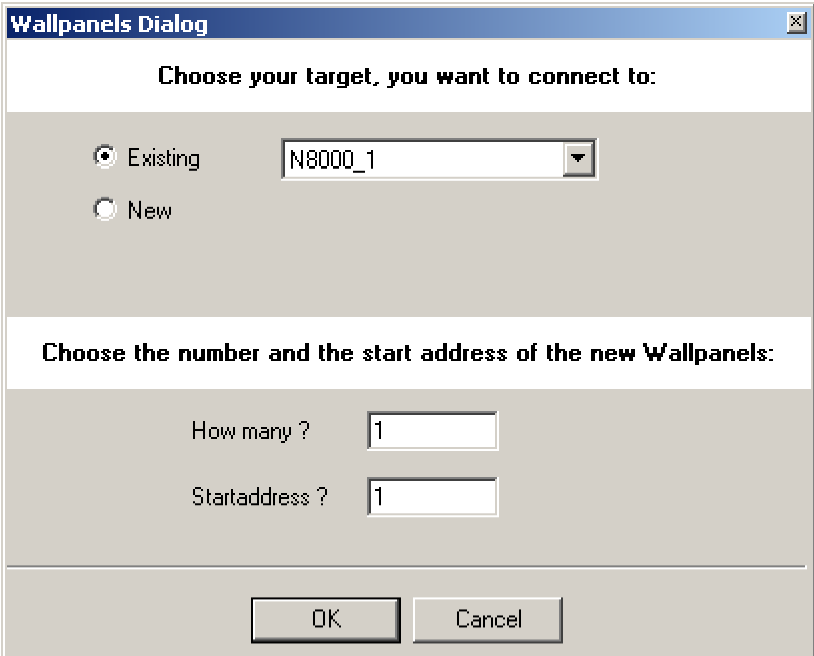
Front user panels are the parts of a wall station that are viewable for the user. A variety of front user panels with different controls and indicators are available. Up to three user panels can be combined in a single wall station using a single CAN-Bus coupler for all of them. The CAN-Bus coupler PWS-C provides connection between the front user panels and the CAN-Bus. The maximum number of devices on a CAN-Bus (e.g. PWS-C, Remote Amplifier, etc.) depends on the employed central control unit. Please refer to the documentation shipped together with your device.

The complete configuration for wall stations needs to be programmed in IRIS-Net, where you define the number and nature of the wall stations employed in your system. It is also here, where you assign the functions to controls and indicators.

The configuration procedure including all different possibilities will be explained in the following chapters. Please refer to the PWS owner's manual in directory /IRIS-Net/Documentation/Wall Station also.

PWS Device

Begin by creating a PWS device in your IRIS-Net project, by dragging a PWS from the Devices category into the Object List or from the Devices window over the Worksheet (see also chapter Devices and Menu Configurations). The following dialog appears:



Specify the desired number of devices and the communications interface and acknowledge your selection with OK. A single or more PWS devices will appear in the worksheet. Devices can be selected (marked) and freely dragged around or placed within the worksheet.
Double clicking on a PWS device opens the user dialog box. For configuration select the entry Configuration from the contextual menu of the PWS.

Operation

A previously configured wall station can be operated on-line as well as off-line under IRIS-Net. Double click on a PWS in the IRIS-Net worksheet to display the wall station’s front panel. All front panel controls can be operated using the mouse. Meters and indicators show the status of corresponding controls or the controlled parameters respectively.

Offline

IRIS-Net allows Offline-Simulation of a configured wall station to check the programming of indicators and controls prior to actually putting the PWS hardware into operation.

Online

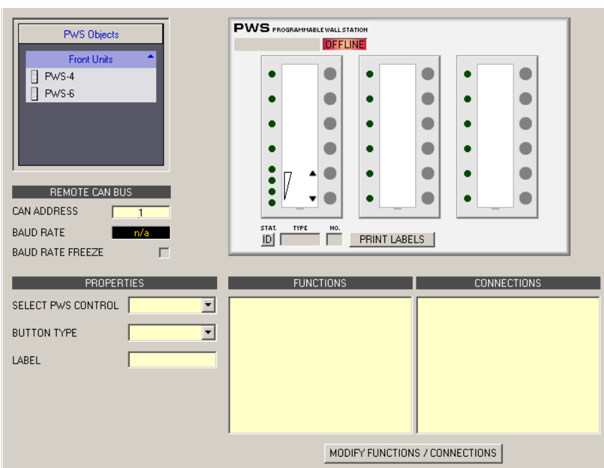
If the wall station is on-line, IRIS-Net shows the actual status of the device’s indicators and controls in real-time. Changes are displayed immediately on the wall station when operating the PWS in IRIS-Net. However, the wall station’s configuration possibilities are limited when in on-line mode.

Configuration

Configuring a wall station is entirely done in the corresponding Configuration Dialog which can be opened by selecting Configuration in the contextual menu of the PWS. A maximum number of up to 3 front user panels can be combined in a single wall station. Assigning any functions to controls and indicators of a front user panel is possible.
Please proceed as follows to configure a wall station:

1. Add the preferred front user panels in the Configuration Dialog at the free positions within the shown triple frame. Add front user panels one by one and from left to right. You can add front user panels either via Drag & Drop operation - out of Front Units of the PWS Objects library or using the entries Add PWS-x of the contextual menu of the corresponding field within the triple-frame.

2. Left-click with the mouse in the desired triple-frame field to select a front user panel for configuration. Use the respective entry in the contextual menu of a front user panel to reposition or delete that panel.
3. Left-click on a front user panel’s control to select it for further configuration. Selecting individual controls via the SELECT PWS CONTROL drop-down menu is possible as well.
4. Assign the desired button type to the selected control using the BUTTON TYPE drop-down menu.
5. Enter the desired name for a control in the LABEL field. (Names can be printed using the PRINT LABELS button)
6. Change the ON VALUE and OFF VALUE parameters of a control, if necessary. Notes on how to manually assign parameter values are provided in the ASCII Control Protocol documentation of the system controller to be controlled.
7. Click on MODIFY FUNCTIONS / CONNECTIONS to set the assigned functionality/parameter of a control. Notes on how to edit function and connections of a control are provided in the chapter IRIS-Net > Editing Objects > Add Control Elements.
   HINT: Up to 12 connections can be assigned to a control.
8. Repeat steps 2 to 7 for all front user panels and their controls.



Element	Description
	Text field for entering a name for each wall station to keep records of its use or position. Click within the grey label field and enter the preferred name and hit Return to accept this name. **CAUTION: The use of * (asterisk) and = (equal) in names is not permissible.**
	The Online / Offline indicator signals the status of the corresponding wall station within the network. The red OFFLINE indicator signals that this wall station is not connected to the network and therefore communication is not possible. OFFLINE also appears when the wall station has not been allocated a network connection in IRIS-Net. Please check the entries in Administrate Connections. The green ONLINE indicator signals that this wall station is connected to the network. Data can be transferred and received. When in on-line mode, all parameter changes are immediately transferred and become immediately active.
	Clicking on the ID buttons lets the LEDs of the wall station’s selected front user panel and the status LED on the PWS-C CAN Bus coupler blink, which facilitates identifying the momentarily accessed PWS, when in on-line mode.

TYPE PWS-4	Type of the selected PWS front user panel.
NO. 1	Number of the selected PWS front user panel. Front user panels are numbered in ascending order if a wall station includes more than one panel of the same type.
PRINT LABELS	Opens a dialog for printing out label field names.

Element	**Description**
REMOTE CAN BUS	
CAN ADDRESS 1	Allows setting a CAN address and indicates the set CAN address. Left-click in the entry field and enter the desired wall station address in the range between 1 and 250. Hit Return to accept the set address. The entered address and the set- ting of the address selection switch on the PWS-C CAN Bus coupler have to be identical and may exist only once on a specific CAN Bus. When adding new wall stations to an IRIS-Net project, CAN addresses are automatically allocated in ascending order.
BAUD RATE	Baud rate of the CAN-Bus to which the wall station is connected. Indication is only provided when in on-line mode. For detailed information on setting the correct baud rate of the System Controller or PWS-C CAN Bus coupler to be cont- rolled, please refer to the according documentation.
PROPERTIES	
SELECT PWS CONTROL	Lets you select a control of the selected PWS front user panel. Selecting the control directly via left-click in the GUI of the front user panel is possible as well.
BUTTON TYPE	Lets you select the desired button type for the selected control. Controls can be used as Action Button (execute action, e.g. .B. loading a preset) Push Button (single push button), Radio Group (a variety of switches that trigger each other – for switching different functions or parameters, e.g. source selection), Switch Button (switch – for switching between two states, e.g. muting) or Channel Selection (only PWS-4, switching parameters edited by up/down buttons).
LABEL	Assign a name to the currently selected control.
ON VALUE/ON VALUES	The control’s state when activated. Notes on how to manually assign parameter values are provided in the ASCII Control Protocol documentation of the system controller to be controlled.
OFF VALUE	The control’s state when deactivated. Notes on how to manually assign parameter values are provided in the ASCII Control Protocol documentation of the system controller to be controlled.
ACTION	Lets you select the action executed when using BUTTON TYPE Action Button.

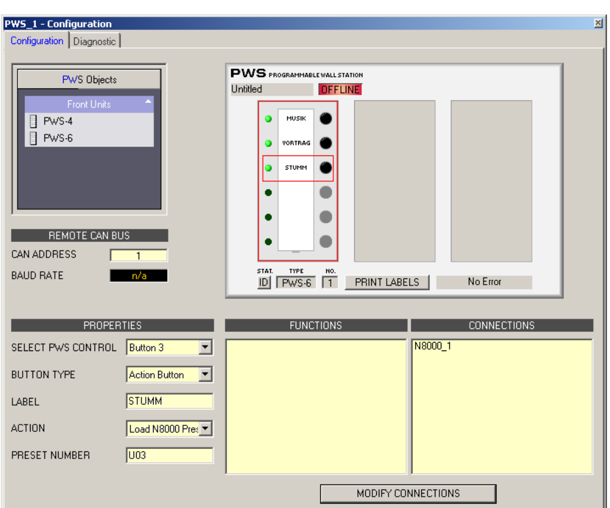
PRESET NUMBER	Lets you select the preset number to be loaded when using BUTTON TYPE Action Button and Action Load N8000 Preset.
LAYERS	Lets you select the number of parameters to be edited by up/down buttons when using BUTTON TYPE Channel Selection.
MODIFY FUNCTIONS / CONNECTIONS	Opens the Modify Functions & Connections dialog for allocating functions to a control. Information on how to allocate Functions & Connections to a control are provided in the chapter IRIS-Net > Editing Objects > Add Control Elements.

Button Types

Action Button

Upon activation (the status changes from OFF to ON), an Action Button carries out a function of a Control. The following example for the usage of the BUTTON TYPE Action Button shows how to configure a PWS-6 to switch between presets in a conference room. Therefore, it is assumed that the IRIS-Net project includes a NetMax N8000 with three presets: MUSIC (U01, background music), SPEECH (U02, e.g. for presentations) and MUTE (U03). The three upper buttons of the PWS-6 shall be configured to switch between these three presets.

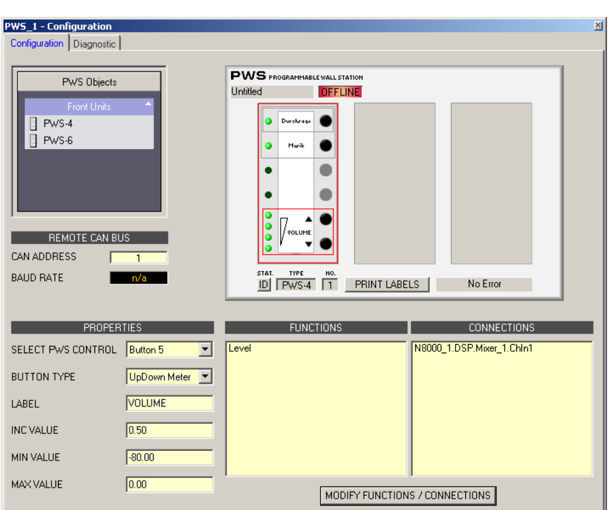
1. Select “Configuration” from the contextual menu of the PWS Device in the IRIS-Net worksheet. The Configuration dialog appears.
2. Add a front unit of the type PWS-6 to the left-hand frame of the three-frame window of the Configuration Dialog. Either use the “drag and drop” method to drag the PWS-6 out of the category Front Units of the PWS objects library or select the entry “Add PWS-6” from the left-hand frame’s contextual menu in the window.
3. Select the first Control on the top of the added PWS-6 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 1” from the SELECT PWS CONTROL dropdown menu.
4. Use the BUTTON TYPE dropdown menu to assign the button type “Action Button” to the selected Control.
5. Name the first preset by entering “MUSIC” into the LABEL input field.
6. Use the ACTION dropdown menu to assign the action-type “Load N8000 Preset” to the selected Control.
7. Number the first preset by entering “U01” into the PRESET NUMBER input field.
8. Click onto the MODIFY CONNECTIONS button and select the N8000.
9. Repeat steps 4 to 9 to assign the preset “U02” to the second Control and the preset “U03” to the third Control as well as to name the presets accordingly. The following illustration shows the Configuration Dialog after configuration of the third button is finished.



Channel Selection

If the Up/Down buttons of a PWS-4 shall serve to set different parameters, switching between parameters is accomplished using the BUTTON TYPE Channel Selection.
The following example for the usage of the BUTTON TYPE Channel Selection shows how to configure the buttons of a PWS-4 so that the Up/Down buttons can be used to individually control the volume setting for announcements and background music. Therefore, it is assumed that the IRIS-Net project includes a NetMax N8000 with a 2x1 monaural mixer. Announcements shall be fed to input 1 and the background music to input 2. The two upper buttons of the PWS-4 are to be used to select the desired signal (i.e. the corresponding mixer input) and the Up/Down buttons can be used to control the volume setting of the selected input channel.

1. Select “Configuration” from the contextual menu of the PWS Device in the IRIS-Net worksheet. The Configuration dialog appears.
2. Add a front unit of the type PWS-4 to the left-hand frame of the three-frame window of the Configuration Dialog. Either use the “drag and drop” method to drag the PWS-4 out of the category Front Units of the PWS objects library or select the entry “Add PWS-4” from the left-hand frame’s contextual menu in the window.
3. Select the first Control on the top of the added PWS-4 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 1” from the SELECT PWS CONTROL dropdown menu.
4. Use the BUTTON TYPE dropdown menu to assign the button type “Channel Selection” to the selected Control. The same button type is assigned to the second Control automatically.
5. Name the button by entering “Durchsage” into the LABEL input field.
6. Select the last Control of the PWS-4 using the left mouse button. As an alternative, select the Control by choosing the entry "Button 5" from the SELECT PWS CONTROL dropdown menu.
7. Use the BUTTON TYPE dropdown menu to assign the button type "UpDown Meter" to the selected Control.
8. Click on button MODIFY FUNCTIONS / CONNECTIONS and select function "Level" and connection "N8000_1.DSP.Mixer_1.ChIn1".
9. Select the second Control on the top of the added PWS-4 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 2” from the SELECT PWS CONTROL dropdown menu.
10. Name the button by entering “Musik” into the LABEL input field.
11. Select the last Control of the PWS-4 using the left mouse button. As an alternative, select the Control by choosing the entry "Button 5" from the SELECT PWS CONTROL dropdown menu.
12. Use the BUTTON TYPE dropdown menu to assign the button type "UpDown Meter" to the selected Control.
13. Click on button MODIFY FUNCTIONS / CONNECTIONS and select function "Level" and connection "N8000_1.DSP.Mixer_1.ChIn2". The following illustration shows the Configuration Dialog after configuration of the button is finished.

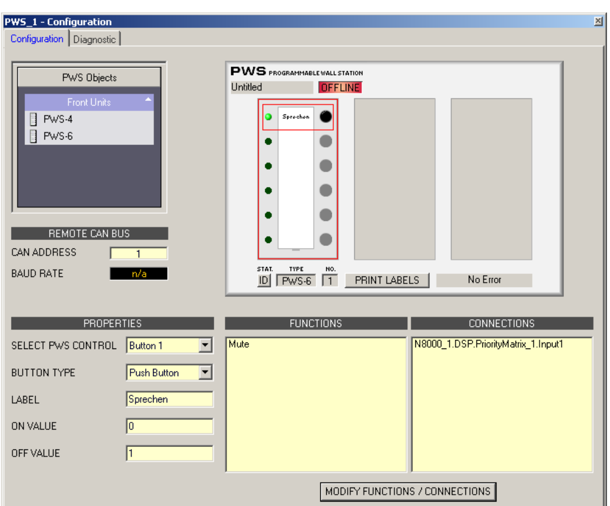


Push Button

A Pushbutton represents a regular button, i.e. it is only activated (ON) as long as the regular button is being pressed. Otherwise, the status of the CONTROL is OFF.

The following example for the usage of the button type "Pushbutton" shows how to configure a button of a PWS-6 to serve as "Push-to-Talk button". Threrefore, it is assumed that the IRIS-Net project includes a NetMax N8000 with a 4 x 4 priority matrix and the signal of a microphone is being fed to input 1 of that matrix. The uppermost button of the PWS-6 shall be used to mute the input channel.

1. Select “Configuration” from the contextual menu of the PWS Device in the IRIS-Net worksheet. The Configuration dialog appears.
2. Add a front unit of the type PWS-6 to the left-hand frame of the three-frame window of the Configuration Dialog. Either use the “drag and drop” method to drag the PWS-6 out of the category Front Units of the PWS objects library or select the entry “Add PWS-6” from the left-hand frame’s contextual menu in the window.
3. Select the first Control on the top of the added PWS-6 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 1” from the SELECT PWS CONTROL dropdown menu.
4. Use the BUTTON TYPE dropdown menu to assign the button type “Push Button” to the selected Control.
5. Name the button by entering “Sprechen” into the LABEL input field.
6. Enter 0 into the ON VALUE input field, so the parameter MUTE is set to 0 (= not muted) when the button is pressed.
7. Enter 1 into the OFF VALUE input field, so the parameter MUTE is set to 1 (= muted) when the button is not pressed.
8. Click on button MODIFY FUNCTIONS / CONNECTIONS and select function "Mute" and connection "N8000_1.DSP.PriorityMatrix_1.Input1". The following illustration shows the Configuration Dialog after configuration of the button is finished.



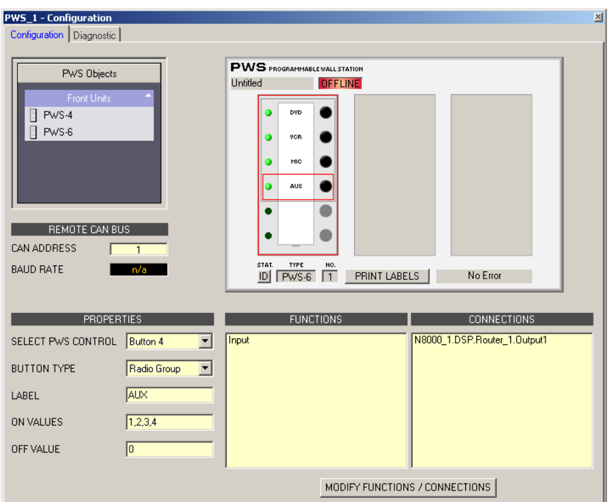
Radio Group

A Radio Group allows switching different values of a parameter. Buttons of a Radio Group disengage each other, i.e. only one button of a group can be activated at a time.

The following example for the usage of the button type "Radio Group" shows how to configure four buttons of a PWS-6 for source selection. Therefore, it is assumed that the IRIS-Net project includes a NetMax N8000 with a 4 x 1 router, with different signal sources being fed to the router’s inputs. The buttons of the PWS-6 shall be used to route exactly one signal source (i.e. the corresponding input on the router) to the router’s output.

1. Select “Configuration” from the contextual menu of the PWS Device in the IRIS-Net worksheet. The Configuration dialog appears.

2. Add a front unit of the type PWS-6 to the left-hand frame of the three-frame window of the Configuration Dialog. Either use the “drag and drop” method to drag the PWS-6 out of the category Front Units of the PWS objects library or select the entry “Add PWS-6” from the left-hand frame’s contextual menu in the window.
3. Select the first Control on the top of the added PWS-6 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 1” from the SELECT PWS CONTROL dropdown menu.
4. Use the BUTTON TYPE dropdown menu to assign the button type “Radio Group” to the selected Control.
5. Enter "1,2,3,4" into the ON VALUE input field. This defines the size of the group as four and at the same time the parameter value set by each of the four radio buttons. The ON VALUE is set automatically for all members of the radio group.
6. Enter 0 into the OFF VALUE input field. This is the parameter to be set when no button is pressed. The OFF VALUE is set automatically for all members of the radio group.
7. Click on button MODIFY FUNCTIONS / CONNECTIONS and select function "Input" and connection "N8000_1.DSP.Router_1.Output1". The function and connection is set automatically for all members of the radio group.
8. Enter the name of the signal at input 1 of the Router into the LABEL input field, e.g. "DVD".
9. Repeat step 9 for buttons 2, 3 and 4. Enter the corresponding signal names (e.g. VCR, MIC, AUX) ein. The following illustration shows the Configuration Dialog after configuration of the radio group is finished.

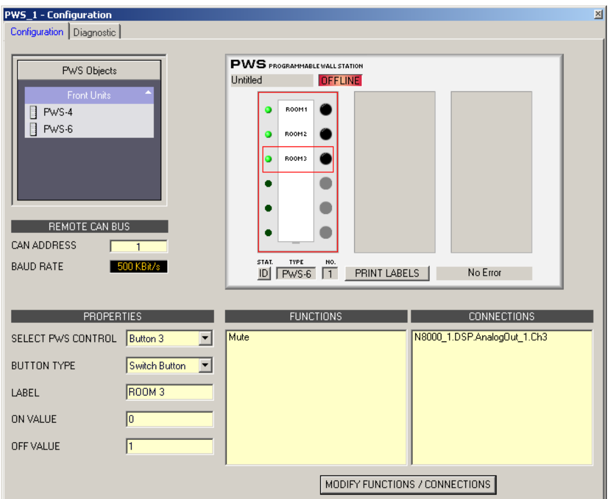


Switch Button

A Switch Button represents a regular switch, i.e. the switch setting changes every time when switching from ON to OFF or from OFF to ON. Otherwise, the current state is remained (latching button). The following example for the usage of the BUTTON TYPE “Switch Button” shows how to configure the buttons of a PWS-6 to be able to select different sound reinforcement zones of a building. Therefore, it is assumed that the IRIS-Net project includes a NetMax N8000 equipped with an AO-1 Analog Output Card, with the card’s outputs 1 to 3 representing the different areas in a building. The upper three buttons of the PWS-6 shall serve to mute the corresponding output signals.

1. Select “Configuration” from the contextual menu of the PWS Device in the IRIS-Net worksheet. The Configuration dialog appears.
2. Add a front unit of the type PWS-6 to the left-hand frame of the three-frame window of the Configuration Dialog. Either use the “drag and drop” method to drag the PWS-6 out of the category Front Units of the PWS objects library or select the entry “Add PWS-6” from the left-hand frame’s contextual menu in the window.
3. Select the first Control on the top of the added PWS-6 using the left mouse button. As an alternative, select the Control by choosing the entry “Button 1” from the SELECT PWS CONTROL dropdown menu.
4. Use the BUTTON TYPE dropdown menu to assign the button type “Radio Group” to the selected Control.
5. Enter „ROOM 1“ into the LABEL input field.

6. Enter 0 into the ON VALUE input field, so the parameter MUTE is set to 0 (= not muted) when the button is pressed.
7. Enter 1 into the OFF VALUE input field, so the parameter MUTE is set to 1 (= muted) when the button is not pressed.
8. Click on button MODIFY FUNCTIONS / CONNECTIONS and select function "Mute" and connection "N8000_1.DSP.AnalogOut_1.Ch1".
9. Repeat steps 4 to 9 for Button 2 (label "ROOM 2", connection "N8000_1.DSP.AnalogOut_1.Ch2") and Button 3 (label "ROOM 3", connection "N8000_1.DSP.AnalogOut_1.Ch3"). The following illustration shows the Configuration Dialog after configuration of the third button is finished.



PROMATRIX 8000

Please note the following when using project files in different versions of IRIS-Net.

“old” IRIS-Net version	“new” IRIS-Net version	Updates
2.2	2.3	The Task Engine of the DPM 8016 has changed, updating all Task Engine blocks is necessary.
2.3	2.4	No updates of project file necessary.
2.4	2.4.1	Level of alarm and chime was increased by 12 dB, the level settings in the project file have to be updated after updating the firmware of the DPM 8016.
2.4.1	2.5	The functionality of the ESC button of the call station has changed in DPC firmware version 1.2.0. Now the ESC button stops the buzzer of the call station only, if a error message appears. The property “FaultMaskGround” of all DPA amplifiers must be set to 0 for correct function of the ground fault indication.
2.5	2.6	No updates of project file necessary.
2.6	2.6.1	The AudioNet dialog of the DPM 8016 has replaced the „PM AudioNet Cfg“ user control. The user control is no longer available. The properties "AudioNetwork.IsMaster", "AudioNetwork.IsSlave" and "AudioNetwork.IsStandalone" are no longer available. The Task Engine checks if more than one assignment to a single element (e.g. relais) is done. In this case an error message appears. Use OR / AND blocks for assigning multiple values to one target.
2.6.1	2.7.1	No updates of project file necessary.
2.7.1	2.7.2	Adjust the detection settings of the new error types „CM-1 PRIMARY LINK“ or „CM-1 SECONDARY LINK“ (see AudioNet dialog of the DPM 8016) to your project needs.
2.7.2	2.8.0	No updates of project file necessary.
2.8.0	2.9.0	No updates of project file necessary.
2.9.0	2.9.1	No updates of project file necessary.
2.9.1	2.9.2	No updates of project file necessary.
2.9.2	2.10.0	The existing property for BGM source selection is no longer used; instead new properties for program assignment have been added. The property is converted into the new properties automatically by IRIS-Net. At the internal message management with PMX-MM-2, some properties have been added. The new properties are added to existing messages automatically.

2.10.0	2.11.0	No updates of project file necessary.
2.11.0	2.11.1	No updates of project file necessary.
2.11.1	3.0.0	No updates of project file necessary.

LIMITS OF THE PROMATRIX 8000 SYSTEM

Property	DPM 8016	System with n DPM 8016 (n_{max} = 10)
Speaker zones, in total	500	1000
– Lines (A/B) per zone	2	
Priority relays, in total	1800	3600
– per zone	5	
Control relays, in total	1800	3600
– per zone	5	
Groups, in total	500	
Call stations per PCA port	4	
Call stations, in total	16	n * 16
Background music program (without audio network)	1	n
Remote control amplifiers per DPM	99	n * 99
Amplifier output channels	396	n * 396
Input modules (with two audio channels each)	8	n * 8
Output modules (with two audio channels each)	8	n * 8
DCS 801R	15	n * 15
DCS 408R or DCS 409R connected to a DCS 801R	17	n * 17
DCS 412R connected to a DCS 801R	5	n * 5
DCS 412R, in total	75	n * 75
Logical inputs, in total 500 n * 500	500	n * 500
DCS 416 connected to a DCS 801	5	n *5
DCS 416, in total	75	n * 75
Supervised inputs from a central fire alarm system	30	n * 30
Relay modules, in total (DCS 408 or DCS 409)	255	n * 255
Number of relays, in total	2000	n * 2000
Task Engine blocks	1024	n * 1024
Task Engine connections per block	48	
Network channels	16	

Property	DPM 8016	System with n DPM 8016 (n_{max} = 10)
Signal trigger	200	n * 200
Internal Alarm generators	2	n * 2
Internal Chime generators	2	n * 2
EOL modules (direct topology only)	250	n * 250
EOL modules per amplifier channel	30	
Remote control connections (e. g. IRIS-Net, DPM 8016)	20	n * 20
Number of MM-2 modules per DPM	2	
Number of message per MM-2	100	
Total length of messages per MM-2	85 minutes	

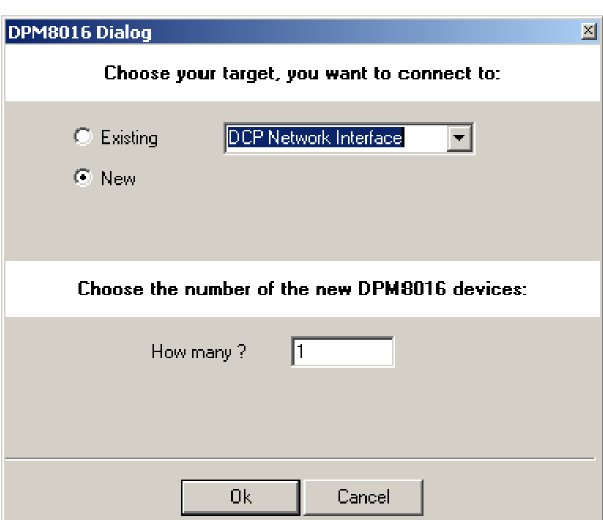
DPM 8016 Paging Manager

The DPM 8016 is the modular, network controlled paging manager of the PROMATRIX 8000 system. Eight audio slots can be assembled with a combination of audio-input and audio-output cards providing a number of flexible configuration options. The DPM 8016 provides all the audio processing, supervision and control functions for a complete PROMATRIX 8000 system. A single DPM 8016 will support up to 16 call stations and 500 paging zones. For larger systems up to 10 DPM 8016 can be interconnected using a digital audio and control network.

Additional DPM 8016 features:
- Integrated audio signal processing - 32 x 16 Routing/Mixer Matrix, EQs, Dynamics, Automatic Gain Control
- Internal supervision with event log - fulfils current requirements of the relevant national and international standards
- Various connection possibilities - Ethernet, CAN-Bus, RS-232, CobraNet, GPIOs

DPM 8016 Device

Start by creating an DPM 8016 Device in your IRIS-Net project. Drag an DPM 8016 from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

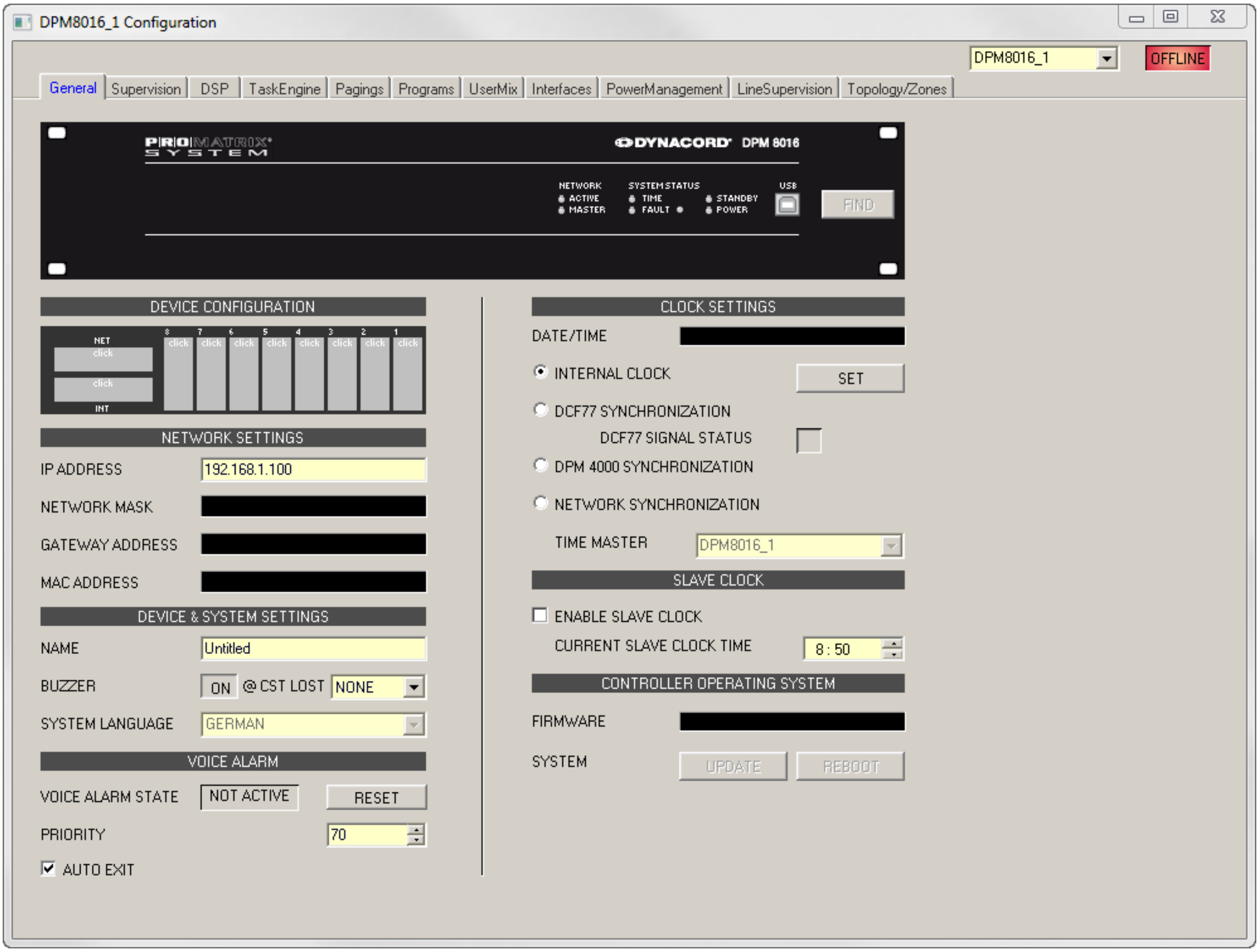


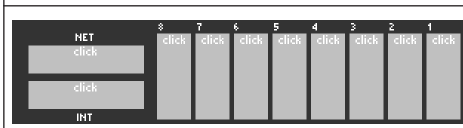
Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.
The following table lists all available device dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	**Description**
General	This window allows hardware settings to be configured, e.g. input/output module slots, network settings, device name, system time and firmware version.
Supervision	This window provides an overview of the operational state and current fault status of the device.
DSP	This window allows editing the DSP configuration of the device.
Task Engine	This window lets you configure the Task Engine of the device.
Pagings	This window lets you configure dynamic add/sub zones (VAR pattern).
User Mix	This window lets you configure background music.
Interface	From this window the DPM 8016 CAN bus, RS-232 ports and GPIO control port interfaces can all be configured. **HINT: Ethernet interface settings are explained under General dialog in the paragraph Network Settings.**
Power Management	From this window the power management of the device can be configured.
Line Supervision	The line supervision of the device can be controlled and supervised from this window.
AudioNet	This window lets you configure the audio network and provides an overview of the current fault status of the audio network. **HINT: This window is only visible if a CM-1 module is included in the configuration.**

General Dialog

Double clicking on a DPM 8016 by default opens the General dialog box. Here, the user can make basic settings that are necessary for flawless operation. All elements of the displayed DPM 8016 front panel are active in on-line mode and correspond to the actual indicators on the unit.

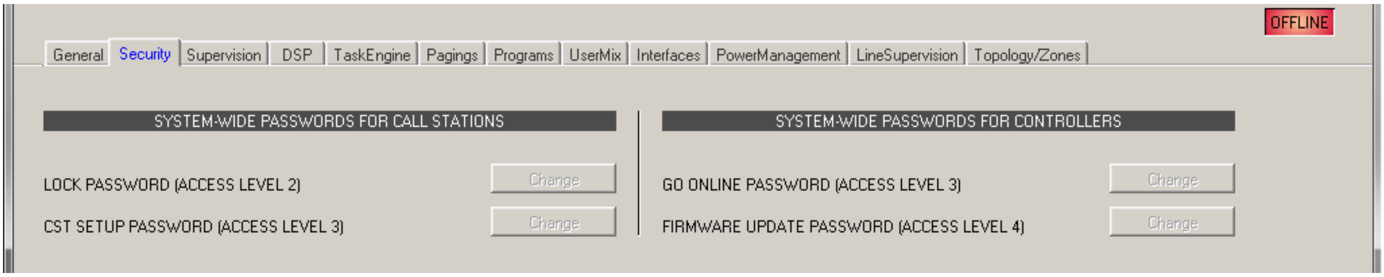


Element	Description
	This view represents the rear panel of a DPM 8016 with module slots and extension cards. In off-line mode, clicking onto the slots with the right mouse button and by exchanging, adding or deleting extension cards, allows defining the unit's configuration. When on-line, the display shows the actually installed extension cards. Differences from the off-line configuration are recognized and marked in yellow or red. **HINT: A yellow indicator signals that the hardware equipment differs from the software configuration. However, this difference does not cause any problems during on-line operation. A red indicator signals an existing conflict between hardware and software configurations, which needs to be remedied, either by customizing the DPM 8016's hardware equipment or through modifying the software configuration.**
IP ADDRESS	Indicates the IP address of the DPM 8016's Ethernet port (factory setting: 192.168.1.100). Enter the address of the DPM 8016 with which you want to establish on-line communication.

NETWORK MASK	Indicates the Ethernet port’s network mask (factory setting: 255.255.255.0).
GATEWAY ADDRESS	Indicates the standard gateway of the Ethernet port (factory setting: 192.168.1.1).
MAC ADDRESS	Indicates the MAC address of the connected DPM 8016 when on-line. The MAC address of the DPM 8016 is also shown on a label on the unit’s rear panel.
NAME	IRIS-Net internal device name of the DPM 8016.
BUZZER	Select ON to indicate a connection failure to a call station (selectable via the drop down field) via the integrated buzzer of the DPM 8016.
SYSTEM LANGUAGE	Select the system language of the PROMATRIX 8000 system.
VOICE ALARM STATE	This indicator shows “ACTIVE” if the device is in voice alarm state, else “NOT ACTIVE”.
RESET	Press the RESET button to deactivate the voice alarm state.
PRIORITY	Select the priority (70–100) of the voice alarm. Select OFF to disable the voice alarm handling of the device.
AUTO EXIT	Select this checkbox if the voice alarm state should be stopped automatically after the alarm signal is stopped/muted (e.g. no alarm request present).
DATE/TIME	Date and time of the DPM 8016 system clock.
INTERNAL CLOCK SET	Opens the system clock settings dialog box.
DCF77 SYNCHRONIZATION	Select this option to synchronize the internal clock of the DPM 8016 with the DCF77 signal. Please refer to the manual how to connect an external DCF77 receiver.
DCF77 SIGNAL STATUS	Indicates the DCF77 signal strength: – Green: Signal strength OK – Red: Signal strength not OK
DPM 4000 SYNCHRONIZATION	Select this option to synchronize the internal clock of the DPM 8016 with a connected DPM 4000 system.
NETWORK SYNCHRONIZATION	Select this option to synchronize the internal clock of this DPM 8016 with the internal clock of another DPM 8016 connected via Ethernet.
TIME MASTER	Select the DPM 8016 (connected via Ethernet) to synchronize the internal clock with. This drop down can only be used if the NETWORK SYNCHRONIZATION option is selected.
ENABLE SLAVE CLOCK	Select this checkbox if slave clocks are connected to the DPM 8016.
CURRENT SLAVE CLOCK TIME	Set the time for the slave clocks.
FIRMWARE	Indicates the firmware version of the DPM 8016 when on-line.
UPDATE	Opens the firmware update dialog.
REBOOT	Reboots the DPM 8016.

Security Dialog

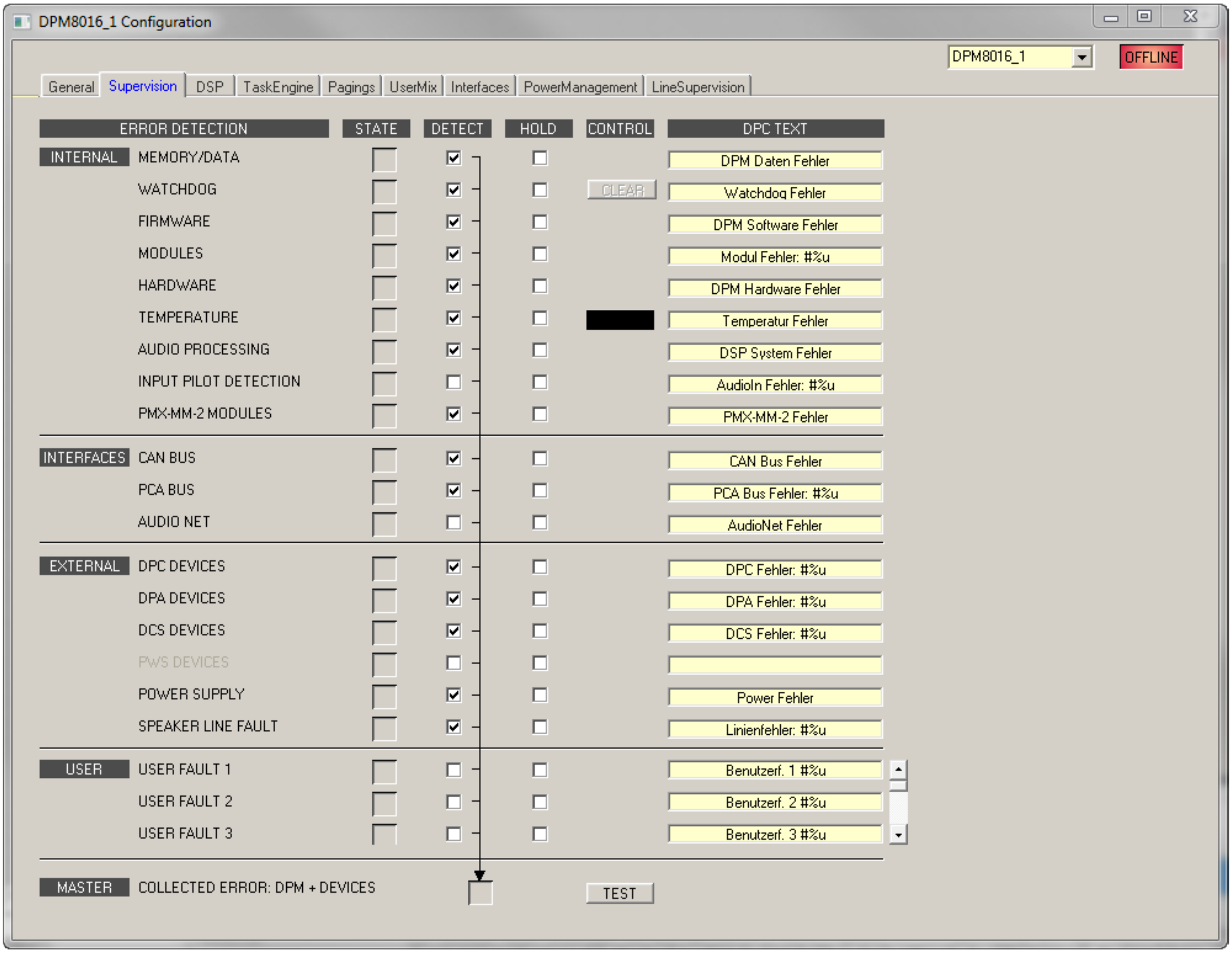
In this dialog the password of the devices can be edited.



Element	Description
LOCK PASSWORD (ACCESS LEVEL 2)	Press the change button to edit the setting of the password for locking call stations.
CST SETUP PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for th setup of call stations.
GO ONLINE PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for going online in IRIS-Net.
FIRMWARE UPDATE PASSWORD (ACCESS LEVEL 4)	Press the change button to edit the setting of the password for updating the firmware of the system.

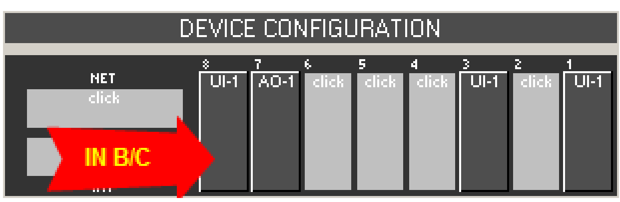
Supervision Dialog

The Supervision window shows the condition of the DPM 8016. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT-LED on the front panel of the DPM lights, the FAULT relay opens and a signal sound.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the corresponding HOLD checkbox is unchecked.
OPEN INTERFACE	Select the checkboxes of the error types to be available via the ASCII Control Protocol of the DPM.
DPC TEXT	If DPC 8015 call stations are configured for error indication, the text entered here is indicated in the call station display if the error occurs. **HINT: The meaning of the parameter %u is described at the error types below.**

INTERNAL

MEMORY/DATA	Memory or Read/Write error.
WATCHDOG	Watchdog error of the DPM. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The DPM firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
MODULES	Invalid module configuration of the DPM 8016. The parameter %u gives the slot number of the invalid module.
HARDWARE	Error in the power supply or the A/D converters of the DPM 8016.
TEMPERATURE	Temperature overload of the DPM 8016.
TEMP	Current temperature on the inside of the enclosure.
AUDIO PROCESSING	Error during the processing of audio data.
INPUT PILOT DETECTION	Pilot tone detection fault at the inputs of the DPM 8016. The parameter %u gives the number of the input. The inputs of the UI-1 modules are numbered ascending. Example:  The parameter %u has the value 6 if there is a detection fault at the IN B/C input of the UI-1 module in slot 8.
PMX-MM-2 MODULE	Error in the PMX-MM-2 module. Please refer to page 525 for details.

INTERFACES

CAN BUS	Fault condition on the CAN bus. Further details are provided in the Interface dialog.
PCA BUS	Fault condition on the PCA bus. Further details are provided in the Interface dialog. The parameter %u gives the slot number of the erroneous module.
AUDIO NET	Fault condition on the audio network.

EXTERNAL

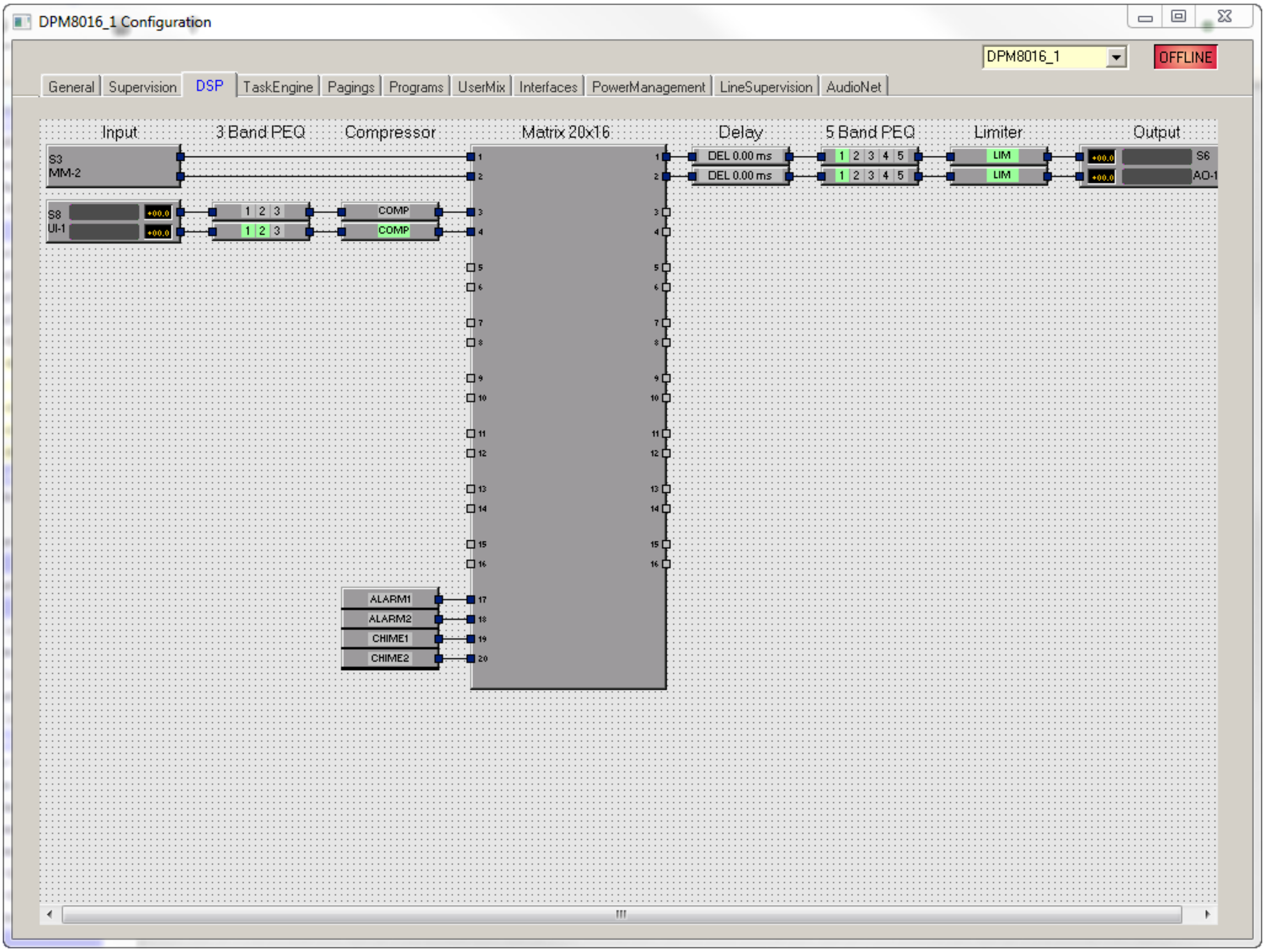
DPC DEVICE	A connected DPC call station has transferred an error message. The parameter %u gives the address of the erroneous call station.
DPA DEVICES	A connected DPA power amplifier has transferred an error message. The parameter %u gives the address of the erroneous amplifier.
DCS DEVICES	A connected DCS system has transferred an error message. The parameter %u gives the address of the erroneous DCS system.
PWS DEVICES	A connected PWS wall station has transferred an error message. The parameter %u gives the address of the erroneous wall station.
POWER SUPPLY	Fault condition in the power supply of the DPM 8016.
SPEAKER LINE FAULT	Fault condition in the speaker line supervision. The parameter %u gives the number of the erroneous speaker line, the number has following meaning: 1 to 500: Zone A 501 to 1000: Zone B

USER

USER FAULT 1 to 10	One or more USER FAULTS have been set **HINT: Use the DPM Task Engine to configure USER FAULTS.**
COLLECTED ERROR	The FAULT-LED on the front panel of the DPM 8016 lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

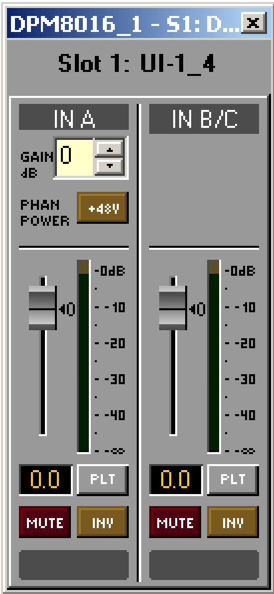
DSP Dialog

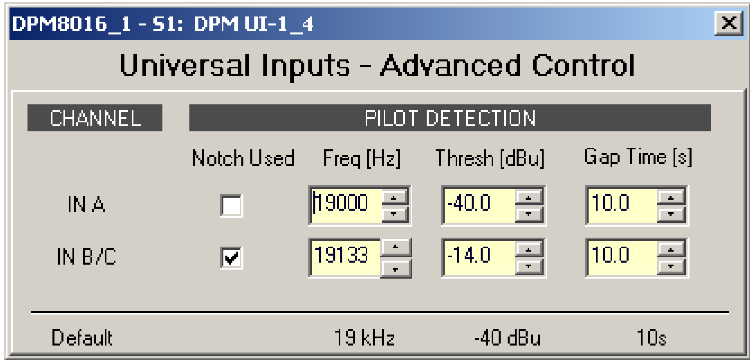
In this dialog the DSP configuration of the DPM 8016 is shown. Double clicking on a DSP-Block’s icon allows editing its configuration and settings in detail.

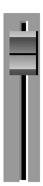


INPUT

The Input block provides access to the inputs of an UI-1 Universal Input Module. The slot number, name and gain values of the input channels are indicated in the block. Double click the block to open the Input dialog. Select the entry Advanced Control from the context menu of the block to open the Advanced Control dialog.





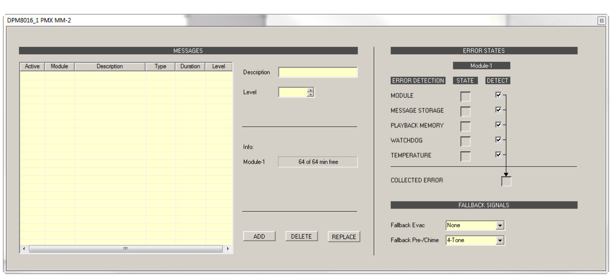
Element	Default	Range	Description
IN A or IN B/C			Permanent channel labeling.
	0.0 dB	0 to 60 dB	The gain of the input channels can be adjusted in 6 dB steps.
			The +48V button is for activating phantom power whenever a suitable condenser microphone is being used.
	0.0 dB	-80 to +18.0 dB	Fader for setting the input level.
	0.0 dB	-80 to +18.0 dB	The fader display shows the numerical value of the current fader setting and additionally allows entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) pilot tone detection. The PLT button lights red when pilot tone detection is active but without a pilot signal being detected. With a pilot signal present, the PLT button lights green.
			MUTE button for muting the input signal.
			INV button for inverting the input signal's polarity.
			Text field for labeling an input channel, e.g. giving it an application specific name. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

Advanced Control

Element	Default	Range	Description
			The checkbox activates a notch filter when pilot tone detection is activated. The notch filter filters an existing pilot tone out of the input signal, so that it will not reach components that are connected behind the input.
	19000 Hz	20 to 20000 Hz	This field sets the frequency of the pilot signal to be detected.
	-40 dBu	-60 to 0.0 dBu	This field sets the pilot tone detection's threshold. The analysis results in OK (green PLT button) when the level of the pilot signal exceeds the threshold. Without a pilot tone being present or if the signal level is below the set threshold, analysis results in a fault message on the corresponding input channel (red PLT button).
	10.0 s		A pilot tone detection error is indicated if the pilot tone is missing longer than the time entered in this field.

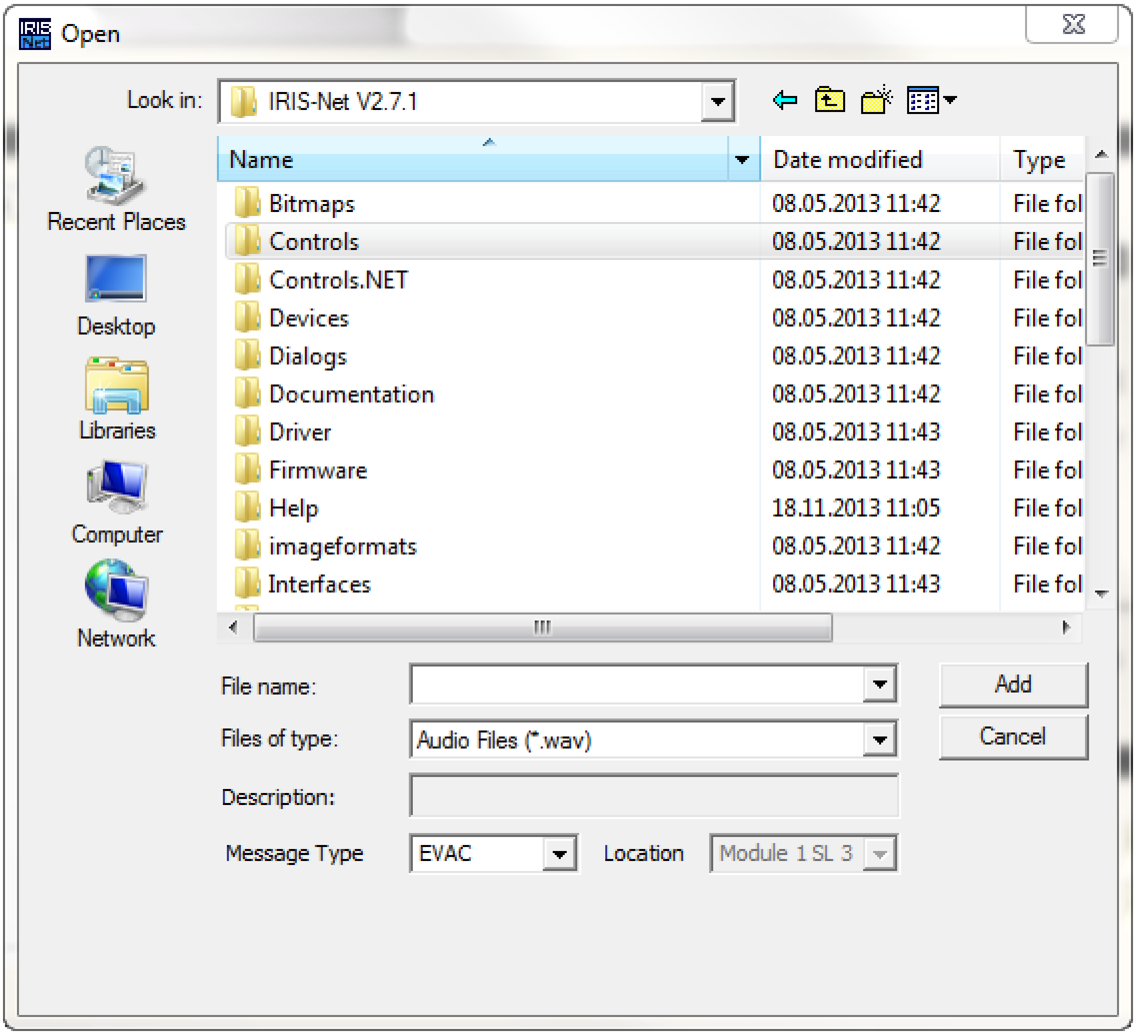
MESSAGE MANAGER MM-2

The MM-2 block provides access to the messages of an MM-2 Message Manager Module. Up to two MM-2 modules can be used in one DPM 8016. The slot number is indicated in the block. Double click the block to open the MM-2 dialog.



Element	Description
MESSAGES	
Active	Indicates the messages that are currently active (marked with a "X").
Module	Gives the number of the module, where the message is stored. The Location can be set when adding messages.
Description	The unique name or description of the uploaded message. Use the corresponding text field to edit the description. The description can be edited in offline or online mode.

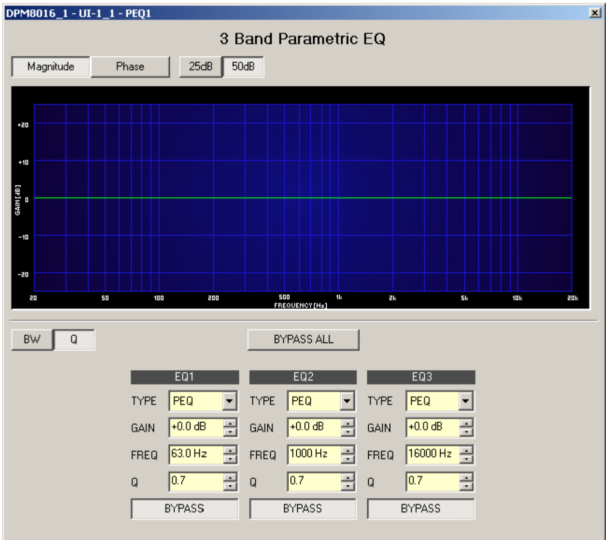
Type	Available message types are EVAC, Chime or Business. The Type can be set when adding messages.
Duration	The duration of the uploaded message, given in format “minutes:seconds”.
Level	Indicates the level of the message. Level ranges from -80 dB to +18 dB. Default level is 0.0 dB. Use the corresponding spin control to edit the level. The level can be edited in offline or online mode.
Info	The memory usage is indicated for all MM-2 modules.
ADD	Press the ADD button to upload a new message. A file selection dialog appears (see screenshot below) that allows selecting a message in WAV file format (mono, 48 kHz). You have to assign a description and a message type (EVAC; Chime or Business) to the message before uploading. If there are two MM-2 modules available, the location for the message has to be selected. **HINT: A selection of standard evacuation messages in different languages is available in the Download area at www.dynacord.com**
DELETE	Press the DELETE button to delete the message selected in the message list.
REPLACE	Press the REPLACE button to replace the message selected in the message list, the message type and location can not be changed. In online mode only business messages can be replaced.
ERROR STATES	
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR flag is set at the same time. Additionally the FAULT-LED on the front panel of the DPM lights, the FAULT relay opens and a signal sounds.
MODULE	Hardware or configuration error in the MM-2 module.
MESSAGE STORAGE	Error during message storage.
PLAYBACK MEMORY	Error in the playback memory.
WATCHDOG	Watchdog error of the DPM. This error type is logged conforming to standards.
TEMPERATURE	Temperature of module is too high.
COLLECTED ERROR	The FAULT-LED on the front panel of the DPM 8016 lights at the occurrence of this type of error.
FALLBACK SIGNALS	
Fallback Evac	Select the default evacuation signal to use if no message is uploaded to the MM-2 module. This settings is valid for all DPM 8016 in the PROMATRIX system.
Fallback Pre-/ Chime	Select the default chime or pre-chime signal to use if no chime is uploaded to the MM-2 module. This settings is valid for all DPM 8016 in the PROMATRIX system.



HINT: For creating audio messages the software Audacity from http://audacity.sourceforge.net/ **can be used.**

3 BAND PEQ

Equalizers accentuate or lower the audio signal within specific frequency ranges. Two parametric 3-Band equalizers are available for each UI-1 Universal Input Module or MM-2 Message Manager Module.

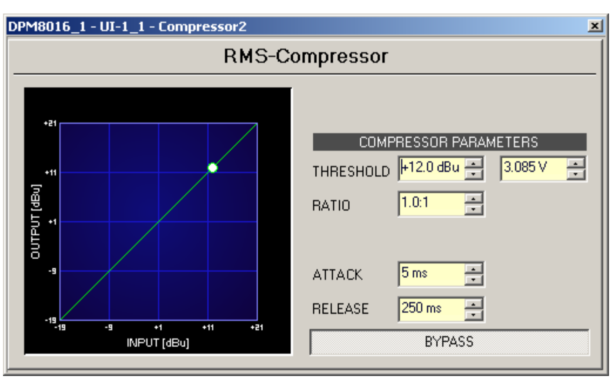


Element	Default	Range	Description
			Switch for displaying amplitude frequency response (magnitude) or phase response (phase)
25dB 50dB			Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
BW Q			Switch for changing between bandwidth BW and Quality Q when setting employed filters.
BYPASS ALL			Pressing BYPASS ALL switches of all filters.
EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope. – Allpass is a filter that has no influence on the frequency response but on the phase response in the transmission function.
GAIN +0.0 dB	0 dB	-18 to +18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.

FREQ 30.0 Hz	63 Hz, 1 kHz, 16 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
BW 1.9 Oct Q 0.7	1.9 Oct or 0.7	0.01 to 6.67 Oct. or 0.1 to 40 (PEQ) 0.1 to 2.0 (Hi-/Lopass)	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

COMPRESSOR

The compressor reduces the dynamic range of audio signals. Once the signal exceeds a certain threshold, the signal gets compressed, i.e. major input level changes result in minor output level changes. Narrowing the dynamic range often allows for easier recording or mixing the audio signal. Two compressors are available for each UI-1 Universal Input Module.

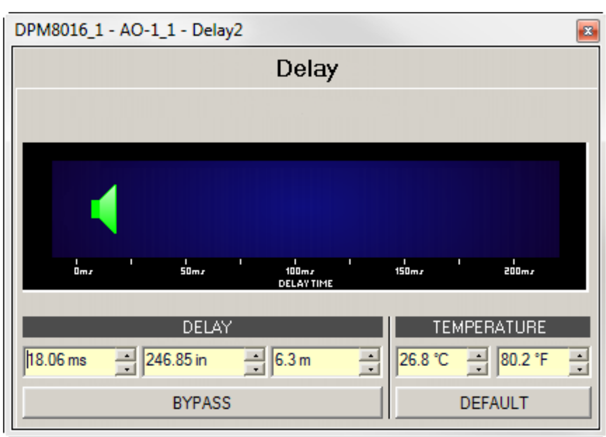


Element	Default	Range	Description
THRESHOLD +0.0 dBu 0.775 V	+12.0 dBu or 3.085 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	THRESHOLD defines the signal level at which the Compressor sets in. Entering the desired value is possible in dBu as well as in V. The entered value is automatically converted in both directions.

	1.0:1	1.0:1 to 8.0:1	RATIO defines the compression rate, i.e. the degree of compression above the threshold level. For example, a rate of 4.0 : 1 represents a signal reduction by factor 4.
	5 ms	0 to 99 ms	ATTACK defines the velocity, at which the compressor sets in. A short attack rate means that even short signal peaks are efficiently com- pressed. Longer attack rate leave signal peaks untouched.
	250 ms	0 to 999 ms	RELEASE defines the control time interval the compressor takes to return to an uncompressed signal level, after the signal dropped below the set threshold.
			BYPASS activates (not engaged) or deactivates the Compressor (engaged), which allows for quick A / B comparison between the com- pressed and uncompressed audio signal.

DELAY

The Delay block allows to delay the audio signal of AO-1 output modules by up to 220 ms.



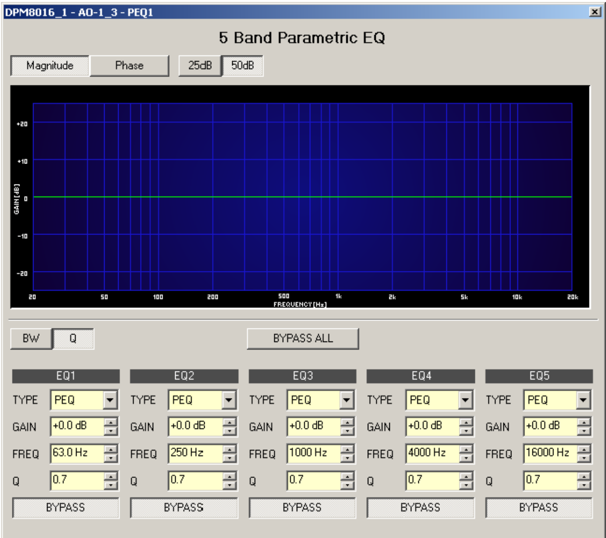
Element	Description
DELAY	Delay times can be entered directly in milliseconds (ms). Alternatively, entering a distance in inches (in) or meters (m) is possible as well. The appropriate delay time is calculated, taking into account the indicated Temperature.
TEMPERATURE	Temperature allows setting the current ambient temperature in degrees Celsius (° C) or degrees Fahrenheit (° F). The two units are automatically converted. The temperature parameter only takes effect if a distance value has previously been entered. In that case, the influence of the temperature is automatically taken into consideration during delay time calculation.
BYPASS	BYPASS either activates (button not engaged) or deactivates (button engaged) the delay which allows for a quick A / B comparison of the effect that set parameters have on the sound characteristics.
DEFAULT	DEFAULT resets the temperature to 20° C or 68° F respectively.

Notice!
The delay block is available for the first 6 output channels (e.g. 3 AO-1 modules) only.

Notice!
Editing Delays by Dragging the Mouse in the Graphics Display
The graphics display shows the speaker symbol in color as soon as a delay has been activated. Clicking with the left mouse button onto the speaker icon and keeping the mouse button pressed allows dragging the symbol to the right or the left, which results in a change of the selected delay time.

5 BAND PEQ

Equalizers accentuate or lower the audio signal within specific frequency ranges. Two parametric 5-Band equalizers are available for each AO-1 Analog Output Module.

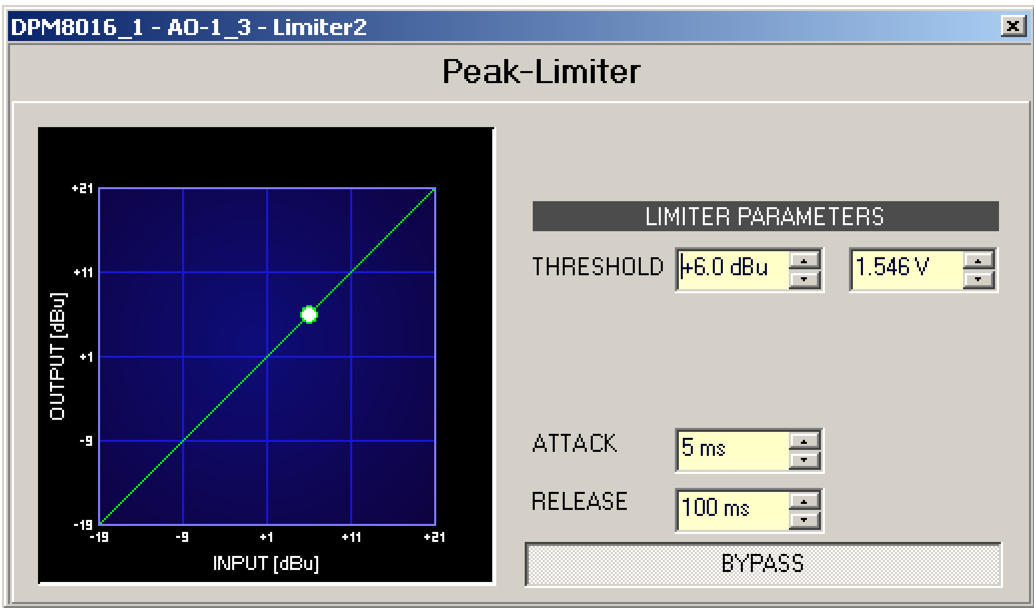


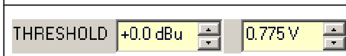
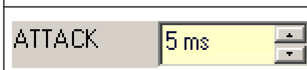
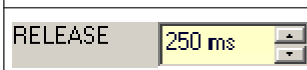
Element	Default	Range	Description
			Switch for displaying amplitude frequency response (magnitude) or phase response (phase)
25dB 50dB			Switch for scaling the amplitude axis to 25 dB (± 12.5 dB) or 50 dB (± 25 dB)
BW Q			Switch for changing between bandwidth BW and Quality Q when setting employed filters.
BYPASS ALL			Pressing BYPASS ALL switches of all filters.
EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass, Allpass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope. – Allpass is a filter that has no influence on the frequency response but on the phase response in the transmission function.
GAIN +0.0 dB	0 dB	-18 to +18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.

FREQ 30.0 Hz	63 Hz, 250 Hz, 1 kHz, 4 kHz, 16 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
BW 1.9 Oct Q 0.7	1.9 Oct or 0.7	0.01 to 6.67 Oct. or 0.1 to 40 (PEQ) 0.1 to 2.0 (Hi-/ Lopass)	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi, Lo and All pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

LIMITER

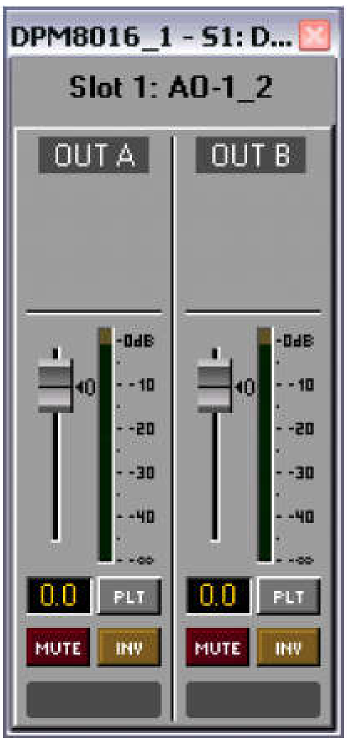
A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage.

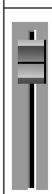


Element	Default	Range	Description
	+6.0 dBu or 1.546 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter unaffected. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Entering the threshold value is possible in dBu or V. The value can be entered in either box and will automatically be converted in the other.
	5 ms	0 to 50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.
	100 ms	10 to 1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

OUTPUT

The Output block provides access to the outputs of an AO-1 Analog Output Module. The slot number, name and gain values of the out channels are indicated in the block. Double click the block to open the Output dialog. Select the entry Advanced Control from the context menu of the block to open the Advanced Control dialog.

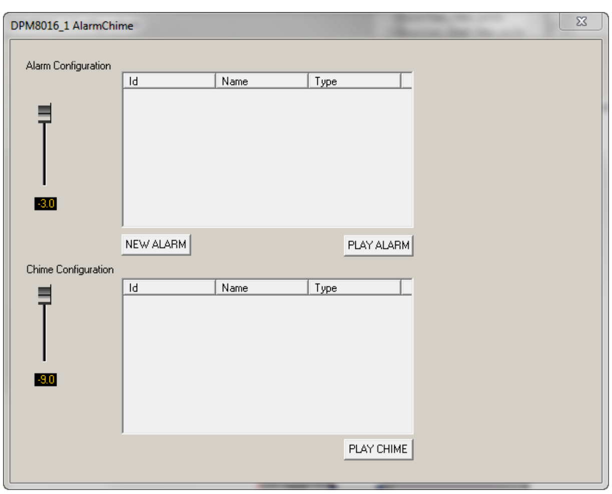


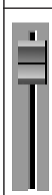
Element	Default	Range	Description
			Permanent channel labeling.
	0.0 dB	-80 to +18.0 dB	Fader for setting the output level.
	0.0 dB	-80 to +18.0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) the pilot tone generator. The PLT button appears only when the pilot tone generator has previously been activated in the Advanced Control window.
			MUTE button for muting the output signal.

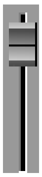
			INV button for inverting the output signal's polarity.
			Text field for labeling an output channel, e.g. giving it an application specific name. **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
HARDWAREJUMPER		6 dBu or 18 dBu	Indicates the jumper setting of the AO-1 module in online mode.
	-45.0 dBu	-60 to 0 dBu	This field allows setting the level of the pilot tone signal.
	19500 Hz	20 to 20000 Hz	This field allows setting the frequency of the pilot tone signal. The set frequency applies to all outputs, for which the pilot tone signal has been activated.

ALARM CHIME

The Alarm Chime dialog allows the configuration of the internal alarm and chime generators.



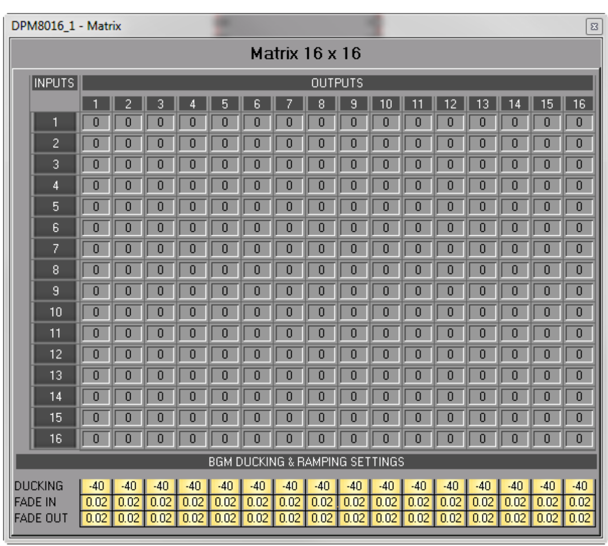
Element	Default	Range	Description
Alarm Configuration			
	-3.0 dB	-80 to 0 dB	Fader for setting the alarm level.

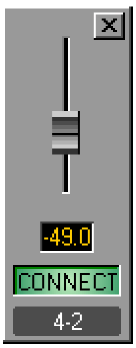
	-3.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
Chime Configuration			
	-9.0 dB	-80 to 0 dB	Fader for setting the chime level.
	-9.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.

MATRIX

Double click on the Matrix 20x16 to open the Matrix 16x16 dialog (the 4 missing inputs in this dialog are used for the internal generators of the DPM 8016). The Matrix 16x16 allows connecting inputs and outputs. Left clicking the node in the matrix where the output channel's column and the input channel's line meet with the mouse does connect an output to an input. Clicking again onto the corresponding node disconnects inputs and outputs.
Please note following restrictions for making connections in the matrix:

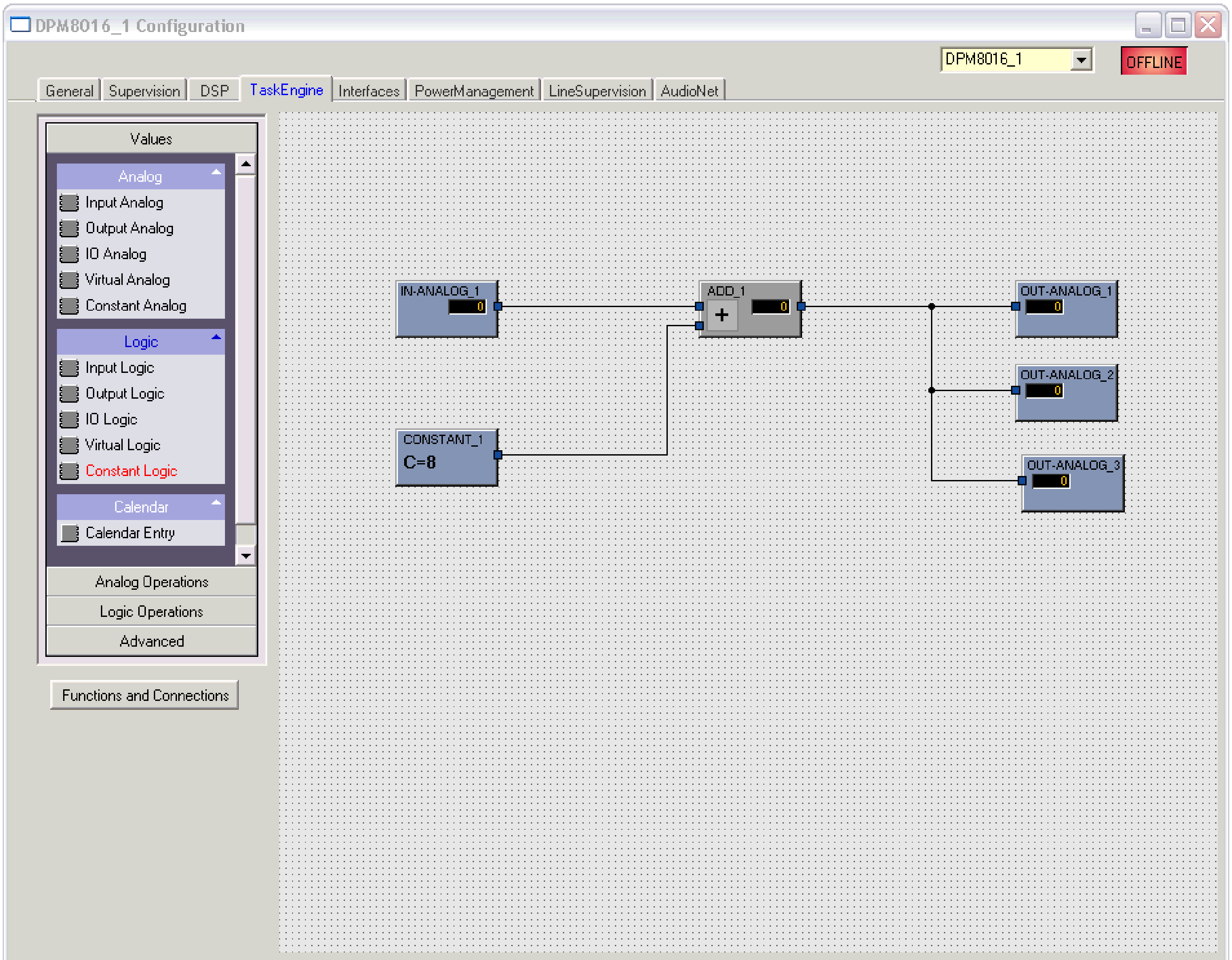
- BGM inputs can only routed via a DPC call station, so this is not possible in this dialog
- Unused inputs can not be routed
- Inputs used for alarms, announcements etc. can not be routed
- Inputs used for a MM-2 Message Module can not be routed
- Manual routings override existing BGM routings
-



Element	Default	Range	Description
DUCKING	-40 dB	-85 to 0 dB	The signal level of the back ground music is reduced by the level entered here when the in input signal, signal level reaches or exceeds a set threshold.
FADE IN	0.02 s	0.01 to 4 s	FADE IN defines how fast the gain of the background music signal is reduced after the input signal exceeds the threshold level.
FADE OUT	0.02 s	0.01 to 0.4 s	FADE OUT defines how fast the gain of the background music signal is returned to the preset level once the input signal drops below the threshold level.
			Right clicking a node opens this dialog with a fader and fader display for setting the level and a CONNECT-Button for setting or resetting the node’s connection. The label represents the node’s position ("line - column") within the matrix. This dialog only appears if a zone is configured for the corresponding output.

Task Engine Dialog

The Task Engine Window allows configuring the Task Engine by dragging inputs, links or outputs from the categories of the FUNCTIONS AND IOS on the left corner of the screen into the Task Engine Worksheet. Elements can be freely positioned and wired within the worksheet. Double clicking on inputs or outputs allows configuring them in detail.
Copy & paste of blocks allow convenient editing of the Task Engine configurations. The size of the worksheet automatically increases when a block is moved to the current border.
Configuring the Task Engine as well as wiring DSP blocks is possible only in offline mode. Please refer to section “How to configure a Control” on page 20 how to assign functions and connections to a Task Engine block.

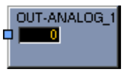
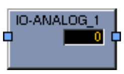
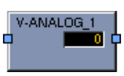
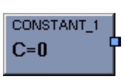
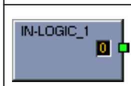
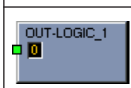
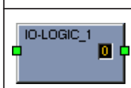


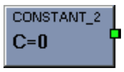
In the Task Engine, one differentiates between two classes of variables:

– Analog: variables of the type "analog" are rational numbers. Example: Level value (-80...+18) of a DSP block mono mixer output.
– Logic: variables of the type "logic" are Boolean values, i.e. only the values "0" and "1" are allowed. Example: Mute (0 = not muted, 1 = muted) of a DSP block monaural mixer output.

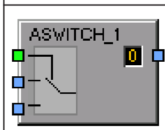
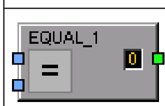
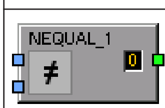
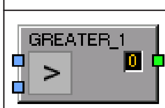
In the Task Engine, different colors are used to distinguish the two types of variables. Inputs and outputs that are not wired are marked blue, whenever variables of the type "analog" are being processed or transmitted. Inputs and outputs that are not wired are marked green, whenever variables of the type "logic" are being processed or transmitted.

VALUES

Element	Description
	The Input Analog block is a variable parameter for rational numbers which always outputs the current value of the connection. The color of the block is yellow if a remote DPM is connected. **HINT: Initialization and updating of Remote Analog Values depends on the connection to the remote DPM 8016.**
	The Output Analog block is a variable parameter for rational numbers which always assigns the current value at the input to the connection. The color of the block is yellow if a remote DPM is connected. **HINT: Initialization and updating of Remote Analog Values depends on the connection to the remote DPM 8016.**
	The IO Analog block is a variable parameter for rational numbers which always outputs the current value of the connection. The current value at the input is assigned to the connection. The color of the block is yellow if a remote DPM is connected. **HINT: Initialization and updating of Remote Analog Values depends on the connection to the remote DPM 8016.**
	The Virtual Analog Value block is similar to the IO Analog block, but has no connection. The keyword Value of the block is used instead of a connection. The keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the DPM. – Persistent = 0: The value is stored in volatile memory.
	The Constant Analog block is a constant parameter for rational numbers. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Input Logic is a variable parameter for Boolean values which always outputs the current value of the connection. The color of the block is yellow if a remote DPM is connected. **HINT: Initialization and updating of Remote Boolean Values depends on the connection to the remote DPM 8016.**
	The Output Logic block is a variable parameter for Boolean values which always assigns the current value at the input to the connection. **HINT: Initialization and updating of Remote Boolean Values depends on the connection to the remote DPM 8016**.
	The IO Logic block is a variable parameter for Boolean values which always outputs the current value of the connection. The current value at the input is assigned to the connection. **HINT: Initialization and updating of Remote Boolean Values depends on the connection to the remote DPM 8016.**

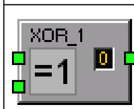
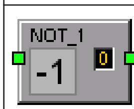
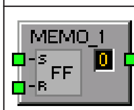
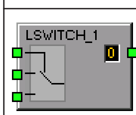
	The Virtual Logic block is similar to the IO Logic block, but has no connection. The keyword Value of the block is used instead of a connection. he keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the DPM. – Persistent = 0: The value is stored in volatile memory.
	The Constant Logic block is a constant parameter for Boolean values. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Calendar Entry block is being used to create time-dependant Boolean values. The outputted Boolean value depends on the configuration of this block and the current system time.

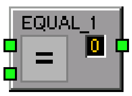
ANALOG OPERATIONS

Element	Description
	The Addition block provides 2 inputs for rational numbers. The rational number at the output is always the sum of rational numbers of the (wired) inputs.
	The Subtraction block subtracts the rational number of the lower input from the rational number of the upper input. The output always presents the result of this analog operation.
	The Multiplication block multiplies the rational number of the upper input with the rational number of the lower input. The output always presents the result of this analog operation.
	The Division block divides the rational number of the upper input by the rational number of the lower input. **CAUTION: If the rational number "0" is present at the lower input, the rational number "0" is always output, independent of the upper input's value.**
	The Switch block switches the rational number at the center or lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Convert block converts a Boolean value to a rational number. The Boolean value 0 is converted to the rational number 0.0, the Boolean value 1 is converted to the rational number 1.0.
	The Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when identical numbers are present at the inputs.
	The Not Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the numbers that are present at the inputs differ from each other.
	The Greater block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than the rational number of the lower input.

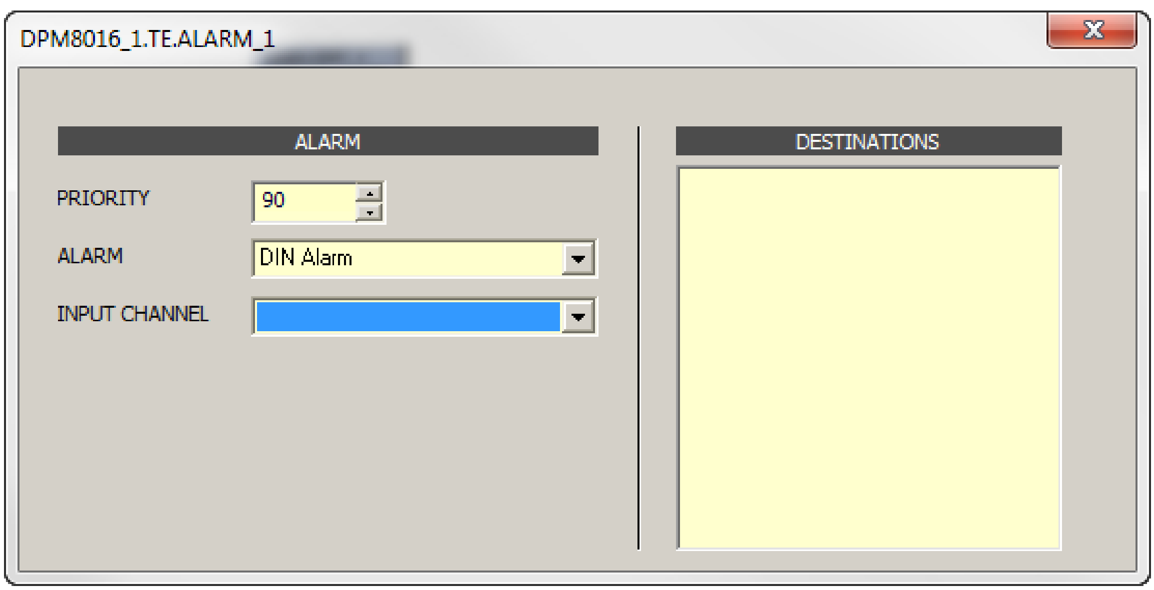
	The Greater Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater or equal than the rational number of the lower input.
	The Less block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower than the rational number of the lower input.
	The Less Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower or equal than the rational number of the lower input.
	The Range Check block allows the range check of a rational number. The Boolean value at the output is only true when the rational number at the input is greater or equal than the property TE.RANGE_CHECK_1.range.min **and** lower or equal than the property TE.RANGE_CHECK_1.range.max – range.max: Enter the upper bound of the range to be checked – range.min: Enter the lower bound of the range to be checked

LOGIC OPERATIONS

Element	Description
	The AND block provides 2 inputs for Boolean values. The Boolean value at the output is only true when all (wired) inputs are true.
	The OR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when at least one (wired) input is true.
	The XOR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when exactly one (wired) input is true.
	The NOT block negates the Boolean value at the input.
	The Memo (Flip-flop) block provides 2 inputs for Boolean values. The upper input sets the flip flop, the lower input resets the flip flop.
	The Switch block switches the Boolean value at the center or the lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Convert block converts a rational number to a Boolean value. The rational number 0.0 is converted to the Boolean value 0, the rational number 1.0 is converted to the Boolean value 1.

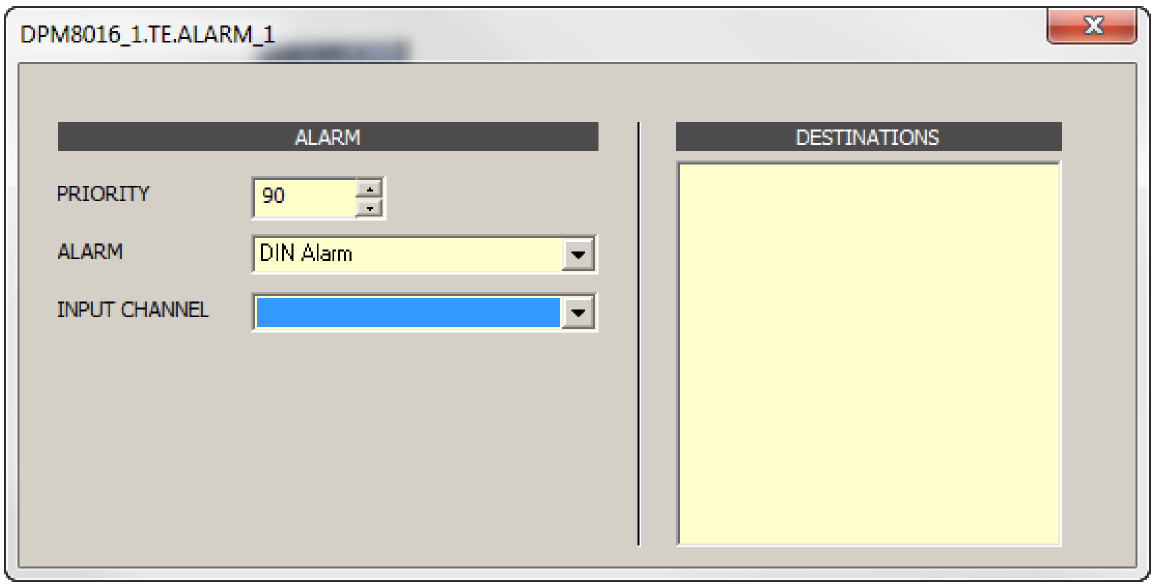
	The Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are identical (e.g. both inputs are true, or both inputs are false).
	The Not Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are different from each other (e.g. one input is true while the other input is false).

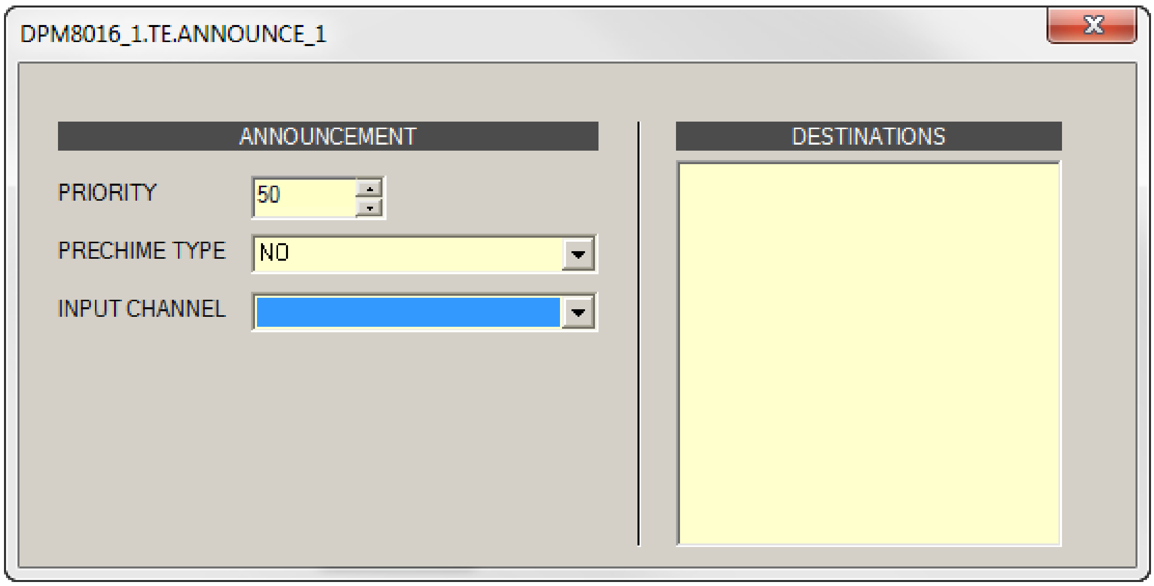
ADVANCED OPERATIONS

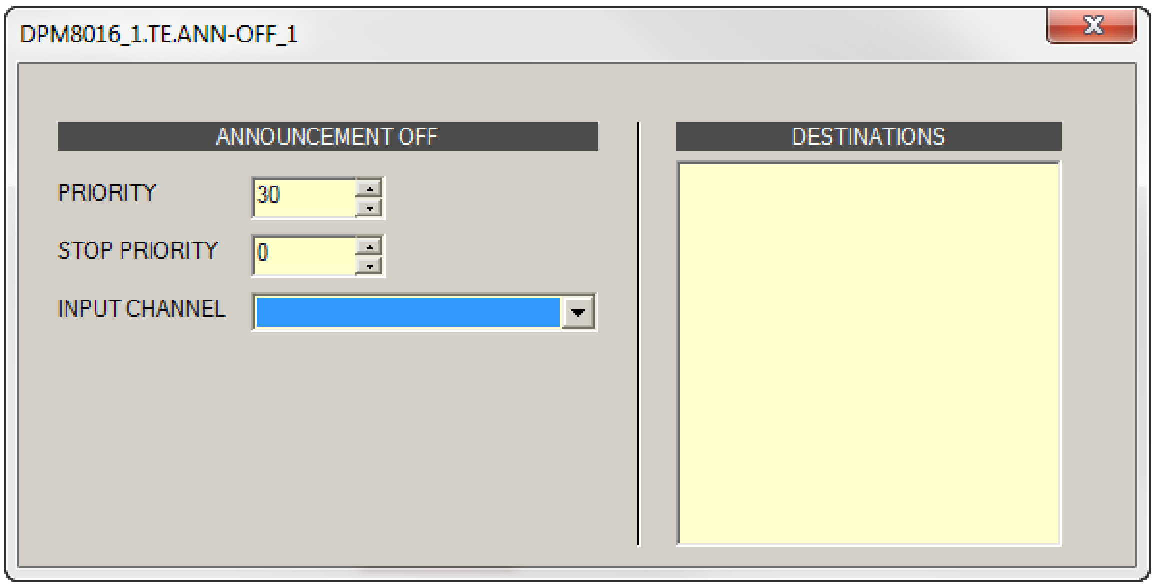
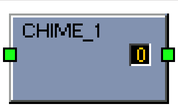
Element	Description
	The Alarm block is used to trigger an alarm. Double click the block to edit the alarm settings (see screenshot below). – PRIORITY: Priority of the alarm (0 to 100). – ALARM: Select the alarm type to be triggered, see table below. – INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the DPM 8016, where the external alarm signal is present. – DESTINATIONS: Select the destination zone(s) or group(s) for the alarm. 

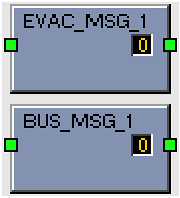
The Manual Alarm is similar to the Alarm block. The additional input T acts like a pushbutton and allows to switch the alarm signal on or off. Double click the block to edit the alarm settings (see screenshot below).

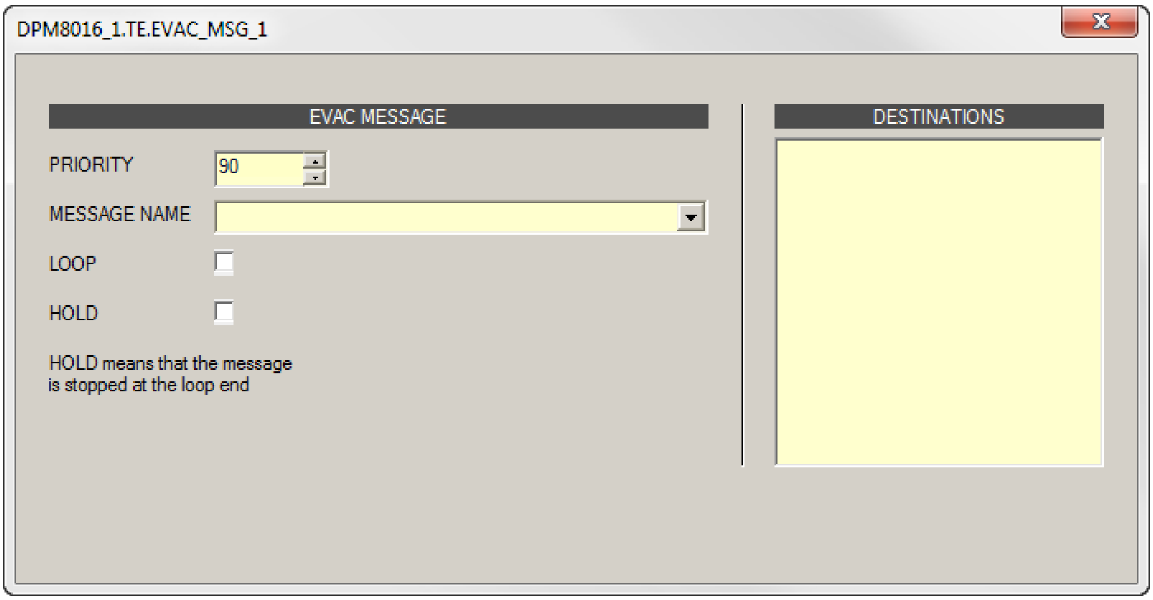
– PRIORITY: Priority of the alarm (0 to 100).
– ALARM: Select the alarm type to be triggered, see table below.
– INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the device, where the external alarm signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s) for the alarm.



	The Announcement block is used to trigger an announcement. Double click the block to edit the announcement settings (see screenshot below). – PRIORITY: Priority of the announcement (0 to 100). – PRECHIME TYPE: Select the pre chime (see table below). Select NO, if there should be now pre chime – INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present. – DESTINATIONS: Select the destination zone(s) or group(s) for the announcement. 

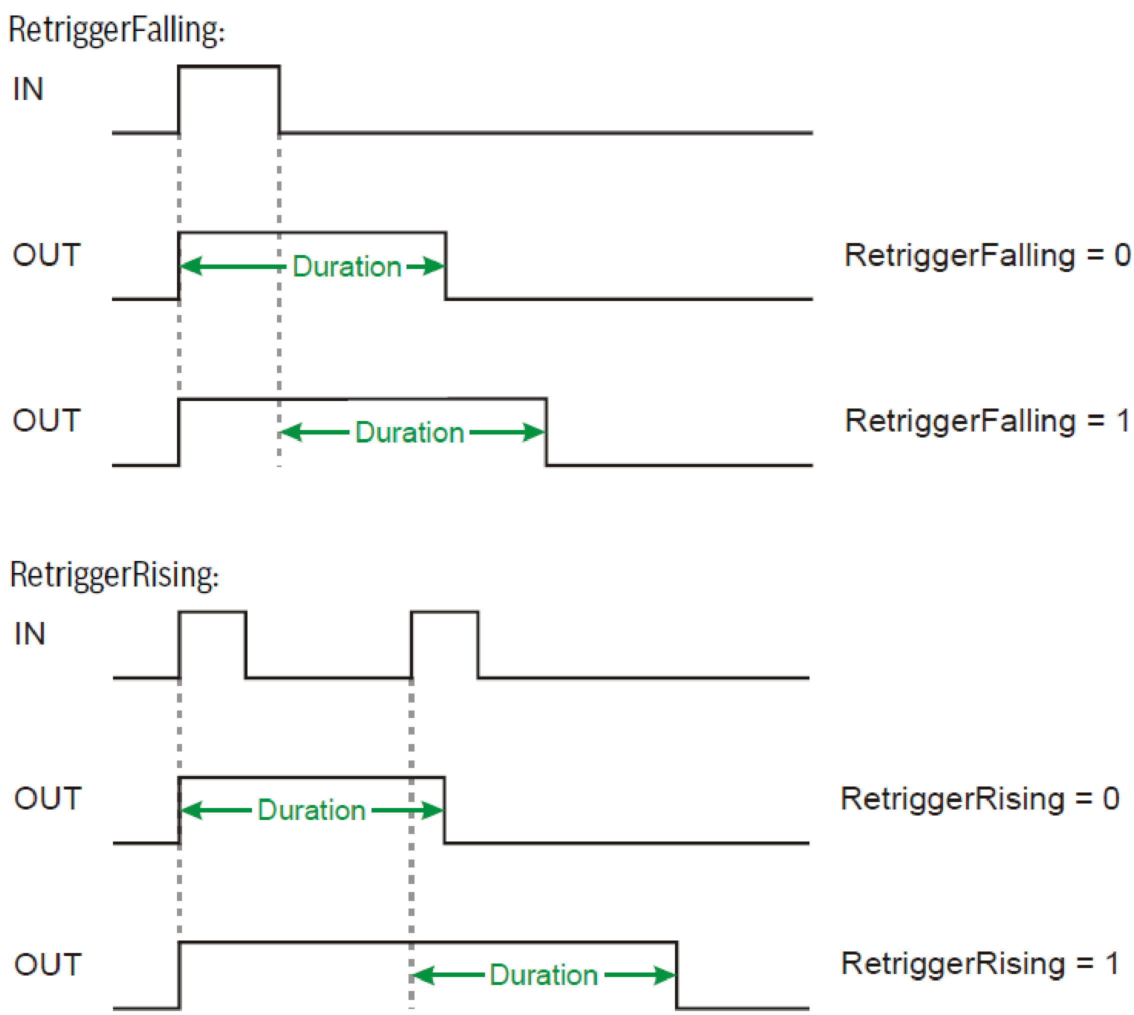
<table>
<tr>
<td></td>
<td>The Announcement OFF block is used to stop an announcement. Double click the block to edit the announcement settings (see screenshot below).
– PRIORITY: Priority of the announcement (0 to 100).
– STOP PRIORITY: Enter the priority (0 to 100) that is used to stop an announcement.
– INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s), where the announcement should be stopped.
</td>
</tr>
<tr>
<td></td>
<td>The Chime block is used to trigger a chime. Double click the block to edit the chime settings.
– PRIORITY: The priority of the chime (0 to 100).
– TYPE: Select the type of the chime.
– LOOP: Select this checkbox to repeat the chime automatically.
– HOLD: Hold means that the message is stopped at the loop end.
– DESTINATIONS: Select the destination zone(s) or group(s) for the chime.</td>
</tr>
</table>

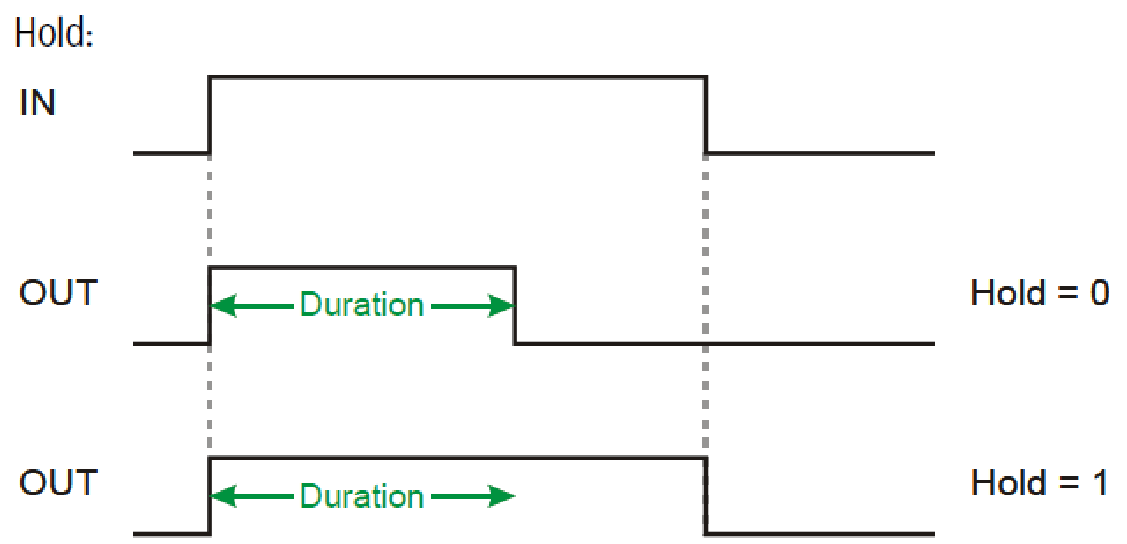
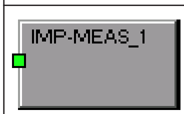
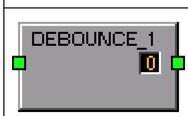
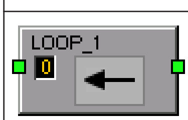
<table>
<tr>
<td>


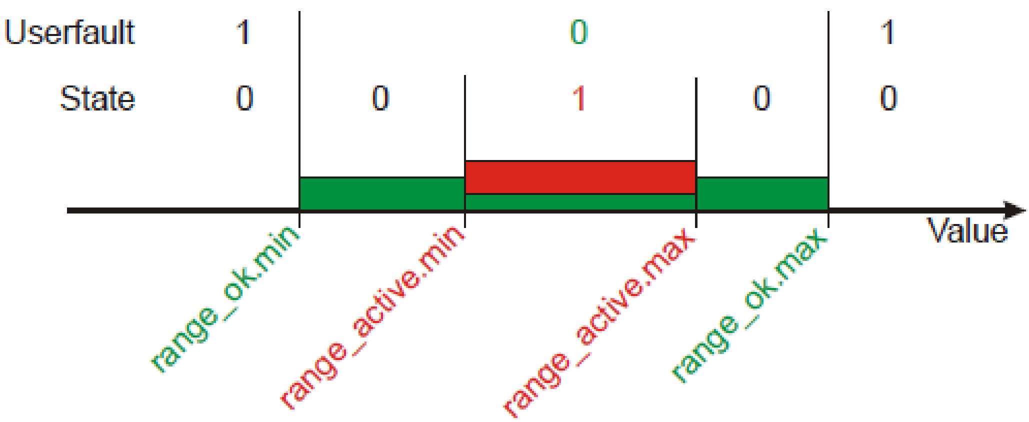
</td>
<td>
The EVAC Message or Business Message block is used to trigger a MM-2 message. Double click the block to edit the message settings (see screenshot below).
– PRIORITY: The priority of the message (0 to 100).
– MESSAGE NAME: Select the (EVAC or Business) message to start.
– LOOP: Select this checkbox to repeat the message automatically.
– HOLD: Hold means that the message is stopped at the loop end.
– DESTINATIONS: Select the destination zone(s) or group(s) for the message.


</td>
</tr>
</table>

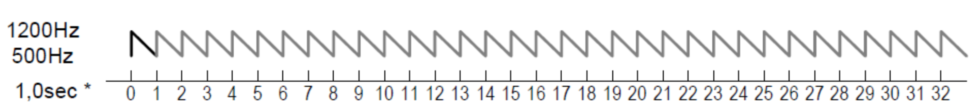
The Timer block sets the state at the output to true for an adjustable duration, when the Boolean value at the input changes from false to true.
- Duration: Enter the duration in seconds, without unit.
- Hold: See illustration below.
- Retrigger Falling: See illustration below.
- Retrigger Rising: See illustration below.
- State: State of the block (1 = time running)
- Timer Value

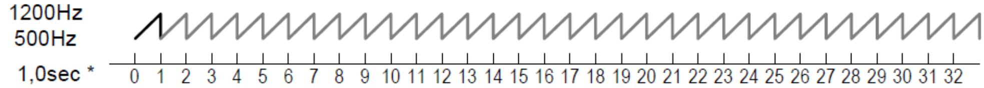
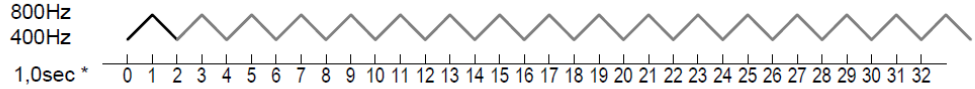
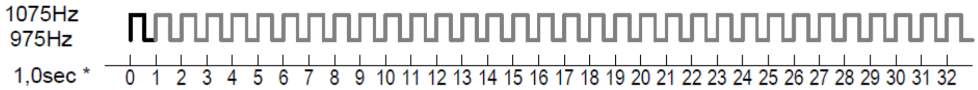
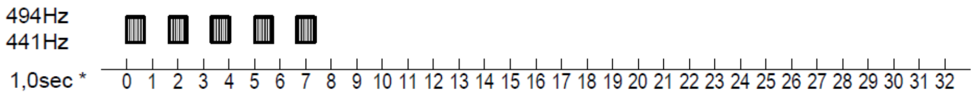
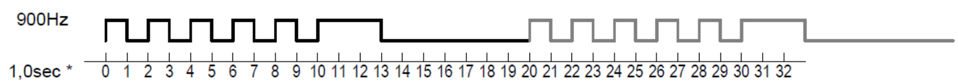
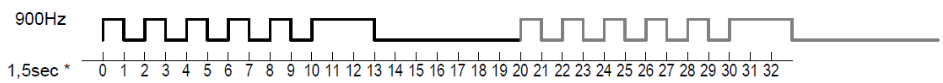
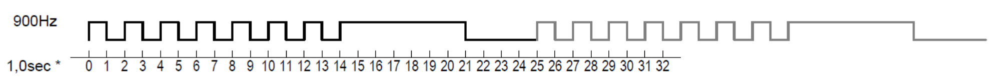
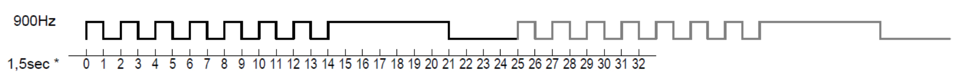
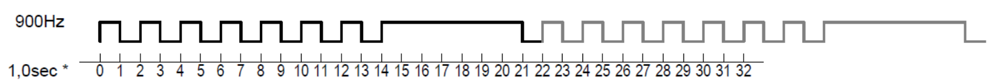
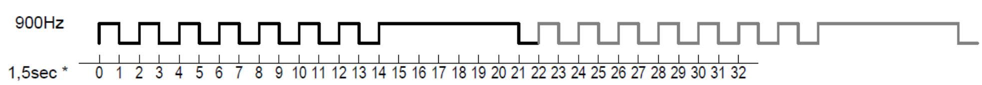
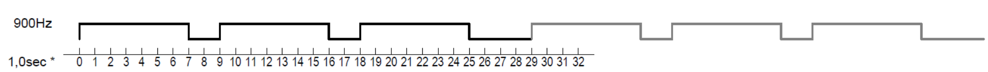
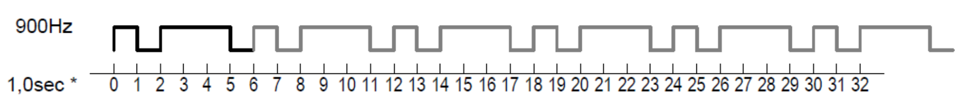
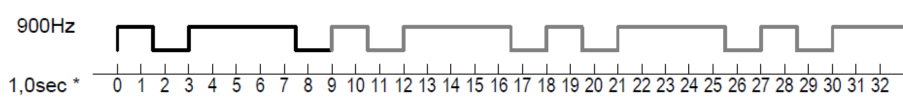


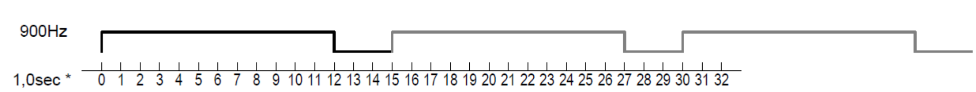
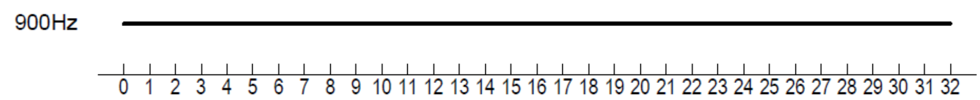
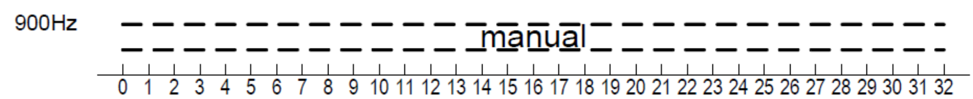
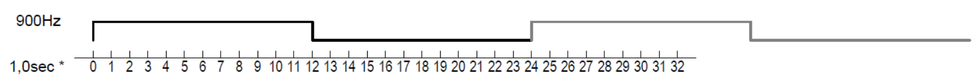
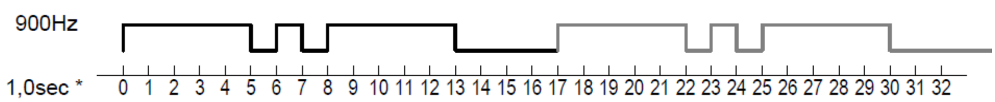
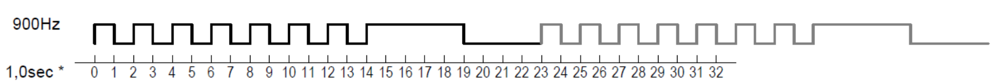
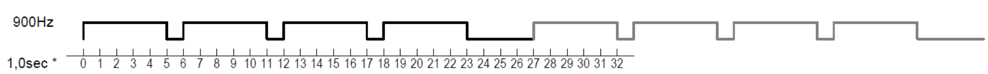
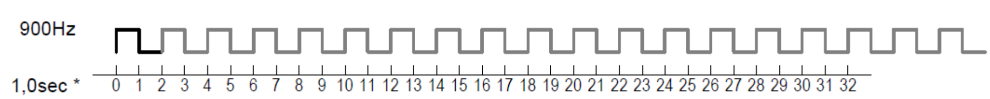
<table>
<tr><td></td><td>
</td></tr>
<tr><td>
</td><td>The CST Text block is used for indicating a text message at the LC display of one or more call stations.
– Acknowledge: Enter 1 if pressing the ESC button at the call station should discard the text at the display.
– Address: Enter the CAN address of the call station, where the text should be indicated. Enter 0 if the text should be indicated at all call stations.
– Buzzer: Enter 1 if the buzzer should signal the text indication additionally.
– Clear: Enter 1 if the text should be cleared when the input changes from true to false.
– Duration: Enter the time in seconds (without unit), how long the text should be indicated.
– State: State of the block (1 = text is indicated)
– Text: Enter the text to be indicated at the display. The maximum length is 20 characters, including space and special characters. See table below for available characters.</td></tr>
<tr><td>
</td><td>The Impedance Measurement block is used for executing a line measurement.
– Lines By Name = ALL
– State: State of the block (1 = measurement active)
– Test Function = LINETEST</td></tr>
<tr><td>
</td><td>The Debounce block is used for debouncing a signal.
– Falling Edge: Enter 1 if the falling edge (change from true to false) at the input should be debounced.
– Rising Edge: Enter 1, if the rising edge (change from false to true) at the input should be debounced.
– State: State of the block
– Time: Enter the debounce time in seconds (without unit)</td></tr>
<tr><td>
</td><td>The Loop block allows building feedback loops in the Task Engine. Unstable conditions are prevented by the block. To point out the function of this block, the input is at the right side, the output is at the left side.</td></tr>
<tr><td>Text Box</td><td>The Text Box allows labeling the task engine configuration. Click the Modify Properties entry in the context menu to open the Edit Textbox dialog. This dialog allows editing the caption and e.g. font size and font type.</td></tr>
</table>

	The Input Supervision block allows supervision of a rational number, especially an input signal from a CIE (Control and Indicating Equipment/fire alarm system). Two ranges can be defined, the Active range and the Ok range. Depending on the ranges the Boolean value at the output (e.g. for triggering an alarm) and a USER FAULT (e.g. for error indication of invalid input values) will be set. The Active range is defined by: – range_active.max: Upper bound of the Active range – range_active.min: Lower bound of the Active range The Boolean value at the output is true if the rational number assigned via Function & Connection is within the Active range. The Boolean value at the output is false if the rational number at the input is below or above the Active range. The Ok range is defined by: – range_ok.max: Upper bound of the Ok range. – range_ok.min: Lower bound of the Ok range. **HINT: If the value of the assigned Function & Connection leaves the Ok range, the State does not change (“state value is latched“)** The USER FAULT is set to 0 if the rational number assigned via Function & Connection is within the Ok range. The USER FAULT is set to 1 if the rational number at the input is below or above the Ok range. Following properties are used to select the USER FAULT: – userfault_connection – userfault_idx 
Superblocks	Superblocks are listed here. Please refer to page 240 how to use Superblocks.

Alarm Types

Type	Graphical Illustration
Extern	-
DIN Alarm	

Slow Whoop	
Siren	
Two-Tone Alarm	
Telephone Alarm	
Ship Alarm 1	
Ship Alarm 2	
Ship Alarm 3	
Ship Alarm 4	
Ship Alarm 5	
Ship Alarm 6	
Ship Alarm 7	
Ship Alarm 8	
Ship Alarm 9	

Ship Alarm 10	
Ship Alarm 11	
Ship Alarm 12	
Ship Alarm 13	
Ship Alarm 14	
Ship Alarm 15	
Ship Alarm 16	
Ship Alarm 17	
SILENCE	-

Chime types

Typ
1_TONE
2_TONE
3_TONE
4_TONE
2x2_TONE
2_TONE_PRE

Characters for DPC text

Available characters for text messages at DPC 8015 display					
!	"	#	$	%	&

<table>
<tr><td>'</td><td>(</td><td>)</td><td>*</td><td>+</td><td>,</td></tr>
<tr><td>-</td><td>.</td><td>/</td><td>0</td><td>1</td><td>2</td></tr>
<tr><td>3</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td></tr>
<tr><td>9</td><td>:</td><td>;</td><td><</td><td>=</td><td>“space”</td></tr>
<tr><td>A</td><td>a</td><td>O</td><td>o</td><td>U</td><td>u</td></tr>
<tr><td>s</td><td>?</td><td>A</td><td>B</td><td>C</td><td>D</td></tr>
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<tr><td>K</td><td>L</td><td>M</td><td>N</td><td>O</td><td>P</td></tr>
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<tr><td>]</td><td>^</td><td>_</td><td>`</td><td>a</td><td>b</td></tr>
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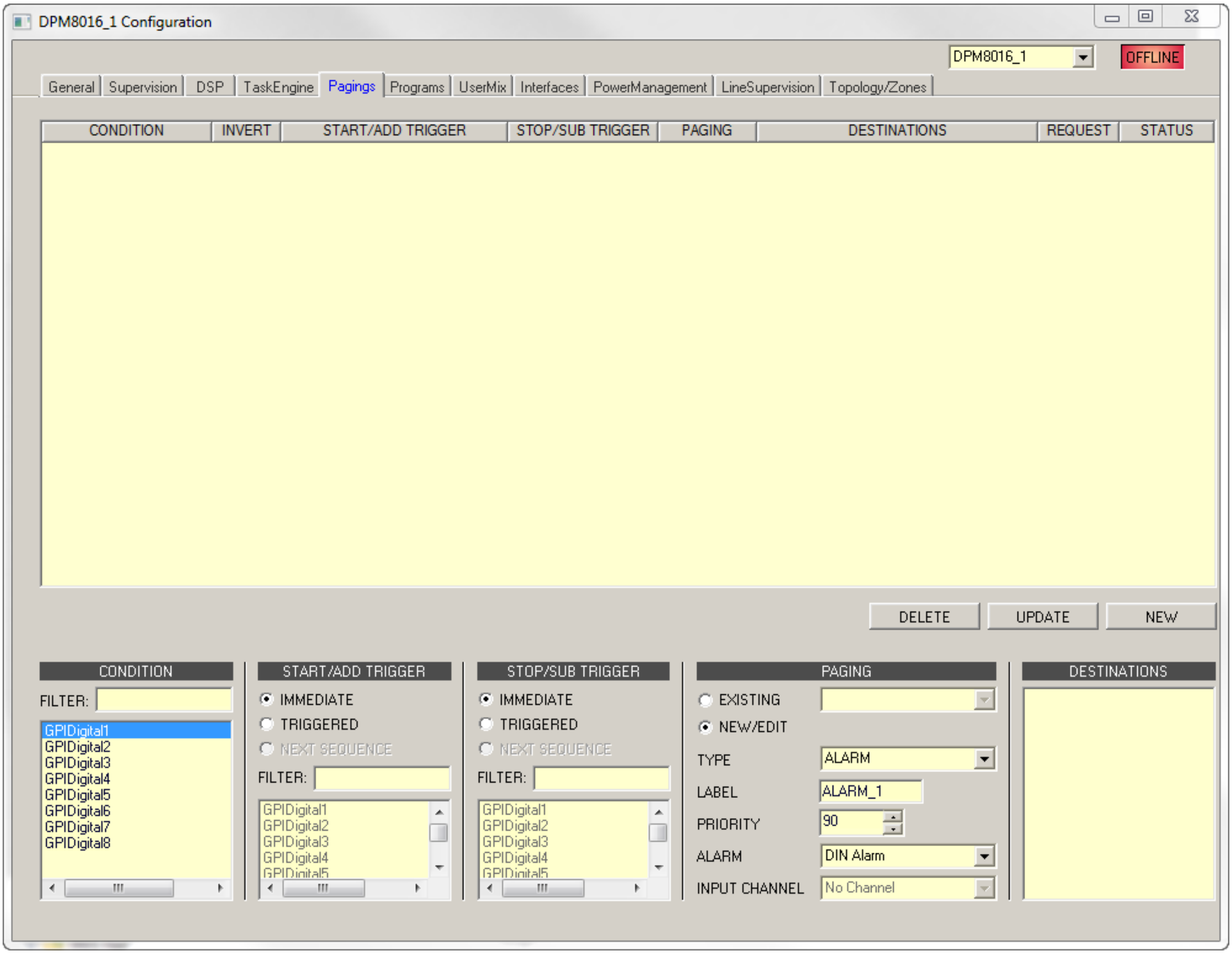
<tr><td>{</td><td>|</td><td>}</td><td>~</td><td></td><td></td></tr>
</table>

Pagings Dialog

The paging dialog allows the configuration of pagings (e.g. alarm or EVAC message) with dynamic destinations.



Element	Description
CONDITION	The status of the condition selected here starts the paging, e.g. the contact of a CIE connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the paging.
START/ADD TRIGGER	The value used to trigger the start of (or addition of destinations to) an active paging. The rising edge of the value is evaluated.
STOP/SUB TRIGGER	The value used to trigger the stop of (or subtraction of destinations from) an active paging. The rising edge of the value is evaluated.
PAGING	The paging initiated by the condition.
DESTINATIONS	The destinations (zones or groups) of the paging.
REQUEST	Indicates if the paging condition is active or inactive.
STATUS	Indicates if the paging is ON or OFF.

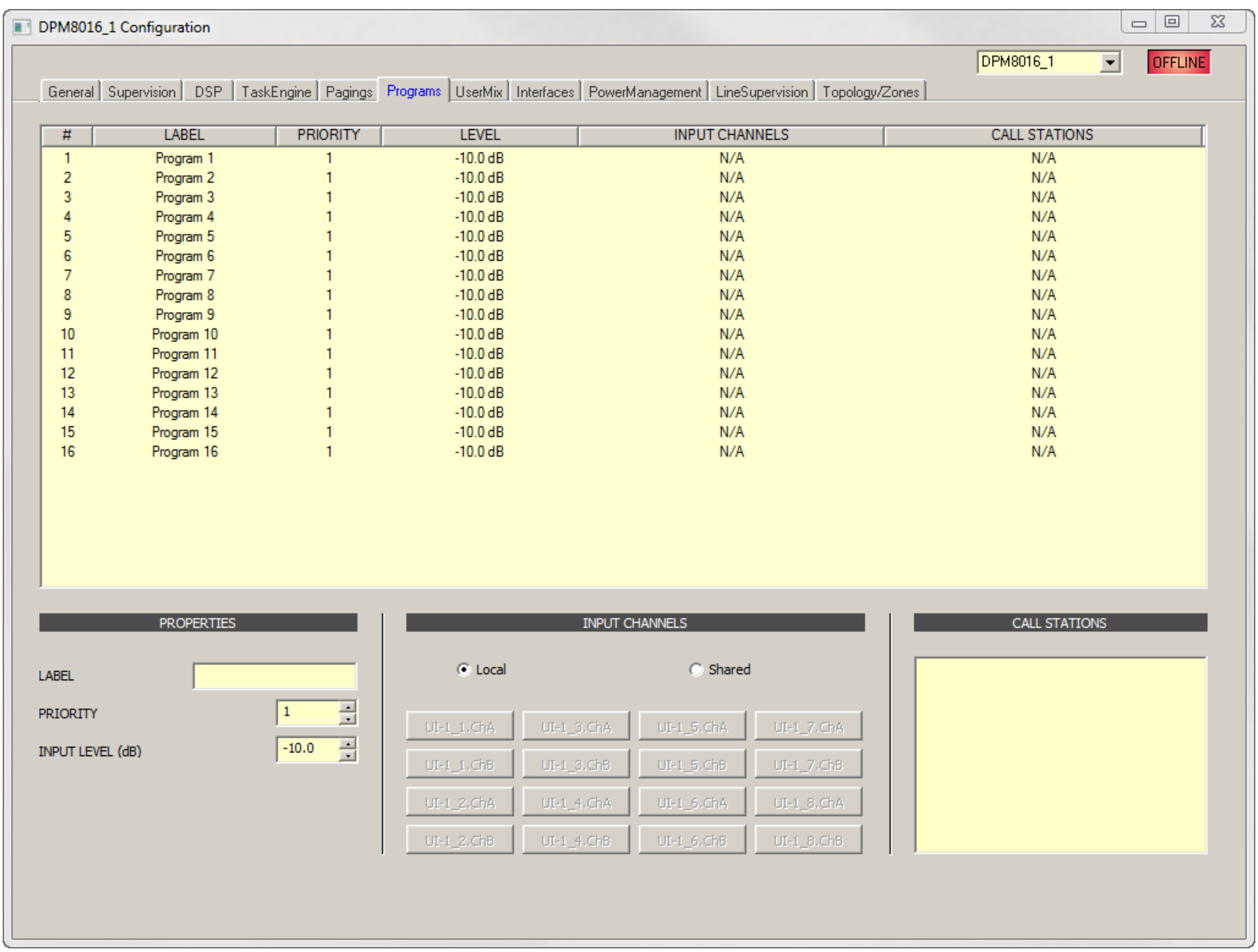
Element	Description
DELETE	Press the DELETE button to delete the paging selected in the paging list.
UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to paging selected in the paging list.
NEW	Press the NEW button to create a new paging using the settings in the lower section of the dialog and adds it to the paging list.
CONDITION	
FILTER and condition list	Select the condition to start a paging from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
START/ADD TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should start immediately or the zones should be added immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be added only after the message ended. When selected, the paging is started immediately. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
STOP/SUB TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should stop immediately or the zones should be removed immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be removed only after the message ended. When selected, the paging is stopped immediately after the message ended. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
PAGING	
EXISTING	Select EXISTING to select an existing paging from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the paging.
TYPE	Select the paging type from the dropdown.
LABEL	Enter the name of the paging.
PRIORITY	Select the priority of the paging.
ALARM	If the selected paging TYPE = ALARM you can select the alarm type from this dropdown.
PRECHIME TYPE	If the selected paging TYPE = ANNOUNCEMENT you can select the prechime type from this dropdown.

CHIME TYPE	If the selected paging TYPE = CHIME you can select the chime type from this dropdown.
MESSAGE NR	If the selected paging TYPE = EVAC Message you can select the message number from this dropdown.
INPUT CHANNEL	If the selected paging TYPE= ANNOUNCEMENT or TYPE = ALARM (and ALARM = Extern) you can select the audio input channel for the paging.
DESTINATIONS	Select the zones or groups of the paging.

Notice!
Setting of routing delay
In large PROMATRIX systems the internal routing of pagings can take quite some time. The property “DPM8000_x.Matrix.Paging.RoutingDelay” allows to delay the activation of the audio signal source for up to 10000 ms. This allows the system to finalize all routing settings and avoids the cut off of pre-chimes or other signals.

Programs Dialog

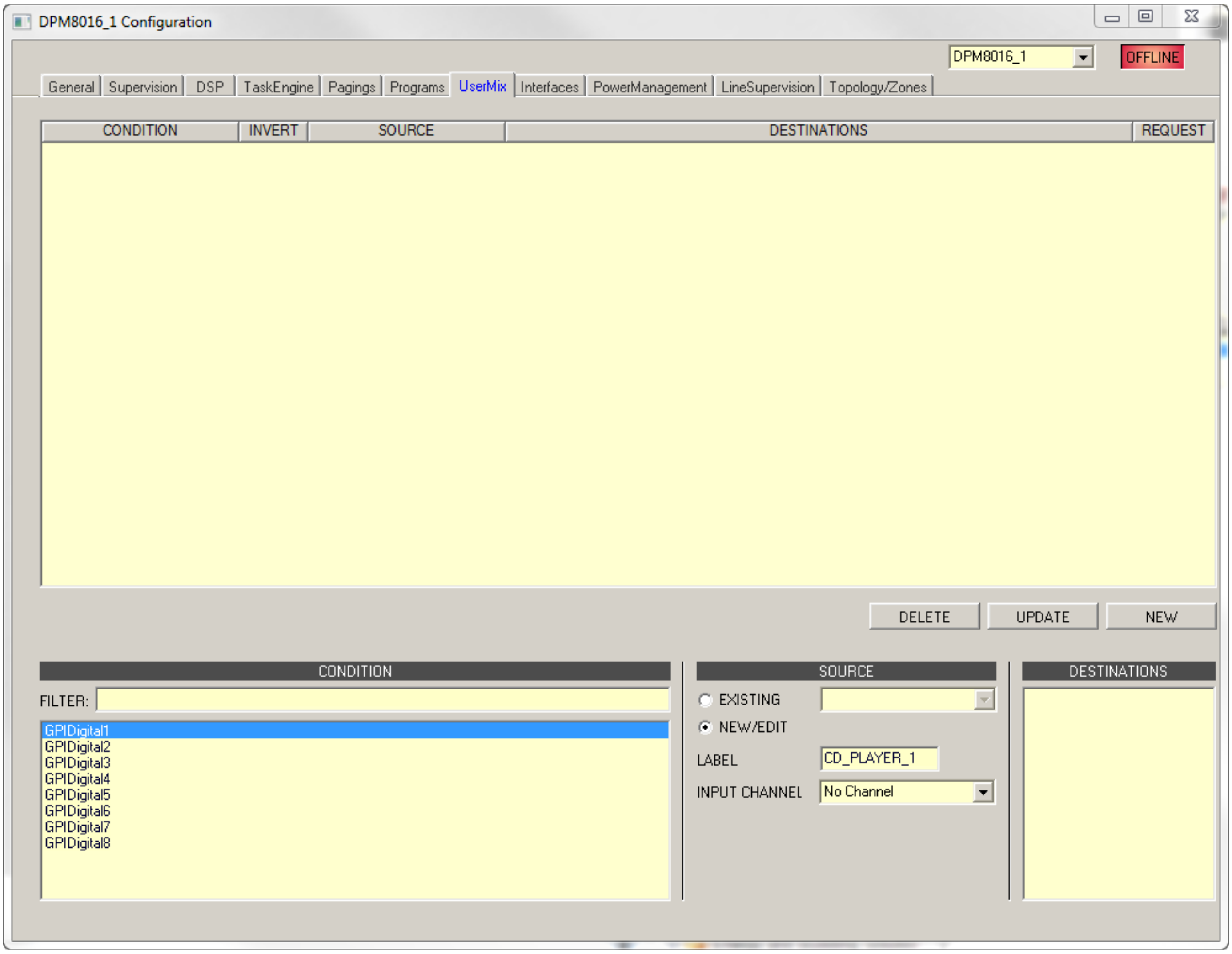
The Programs dialog allows to configure 16 programs for back ground music.



Element	Description
#	Number of the program.
LABEL	Name of the program.
PRIORITY	The priority of the program.
LEVEL	Level of the program.
INPUT CHANNELS	Input channel of the program. Select more than one input channel to mix the audio signals.
CALL STATIONS	The call stations where this program is listed in the menu and can be selected by the call station user.
LABEL	Text field for labeling a program (max. 20 characters), e.g. giving it an application specific name. Note: Using “,” (comma) in a name is not permissible.
PRIORITY	Edit the priority of the program selected in the program list (range: 1 to 69).
LEVEL (dB)	Edit the level of the program selected in the program list (range: -80 to 0 dB). Only the level can be edited in online mode.
Local	Select this option to use one or more local inputs of UI-1 modules (channel A or B) as audio source of the selected program.
Shared	Select this option to use an existing program of another DPM (connected via Ethernet) as audio source of the selected program.
CALL STATIONS	Select the call stations where the selected program will be listed in the menu.

UserMix Dialog

The UserMix window allows configuring audio routings (e.g. background music) in the PROMATRIX 8000 system.



Element	Description
CONDITION	The condition that starts the background music, e.g. a switch connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the back ground music.
SOURCE	The source of the background music.
DESTINATIONS	The destinations (zones or groups) of the background music.
REQUEST	Indicates the current status (active or inactive)

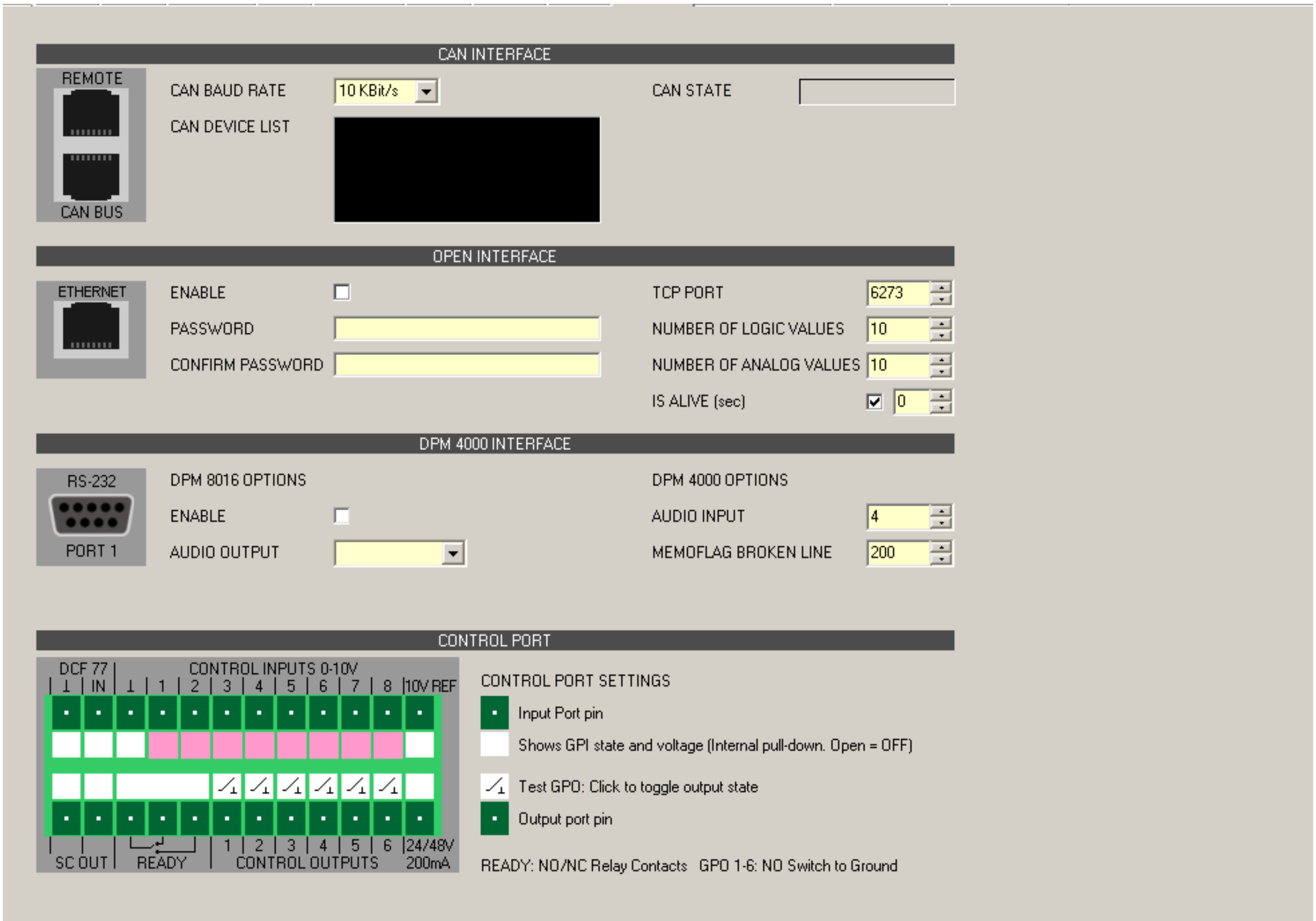
Element	Description
DELETE	Press the DELETE button to delete the entry selected in the list.

UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to entry selected in the list.
NEW	Press the NEW button to create a new background music using the settings in the lower section of the dialog and adds it to the list.

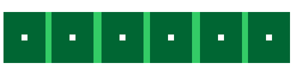
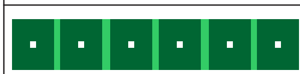
Element	**Description**
CONDITION	
FILTER and condition list	Select the condition to start background music from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
SOURCE	
EXISTING	Select EXISTING to select an existing source for background music from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the source.
LABEL	Enter the name of the background music.
INPUT CHANNEL	Select the audio input channel for the background music.
DESTINATIONS	Select the zones or groups of the background music.

Interface Dialog

The Interface window allows configuring the different interfaces located on the rear panel of the DPM 8016. All REMOTE CAN BUS and DPM 8016 CONTROL PORT settings can be made in here. Configuring the Ethernet interface is done under Network Settings in the General window.



Element	Description
CAN INTERFACE	
CAN BAUD RATE	Transmission rate of the CAN-Bus. All devices on the CAN-Bus must be set to one common transmission rate. **HINT: Editing the CAN BAUD RATE setting is possible in offline mode only.**
CAN STATE	Displays the current CAN-Bus status. Possible indications are: BUS OK, Bus Heavy, Bus Off.
CAN DEVICE LIST	Opens the dialog box for configuring the connected devices.
OPEN INTERFACE	
ENABLE	Set the checkbox to activate the ASCII control protocol of the DPM.
PASSWORD	If password protection of the ASCII control protocol is required, enter the password here. Repeat the password in the CONFIRM PASSWORD field. Go online (write) to set the password in the DPM. **HINT: Editing the password setting is possible in offline mode only.**
TCP Port	TCP port of the ASCII control protocol. The default port is 6273.

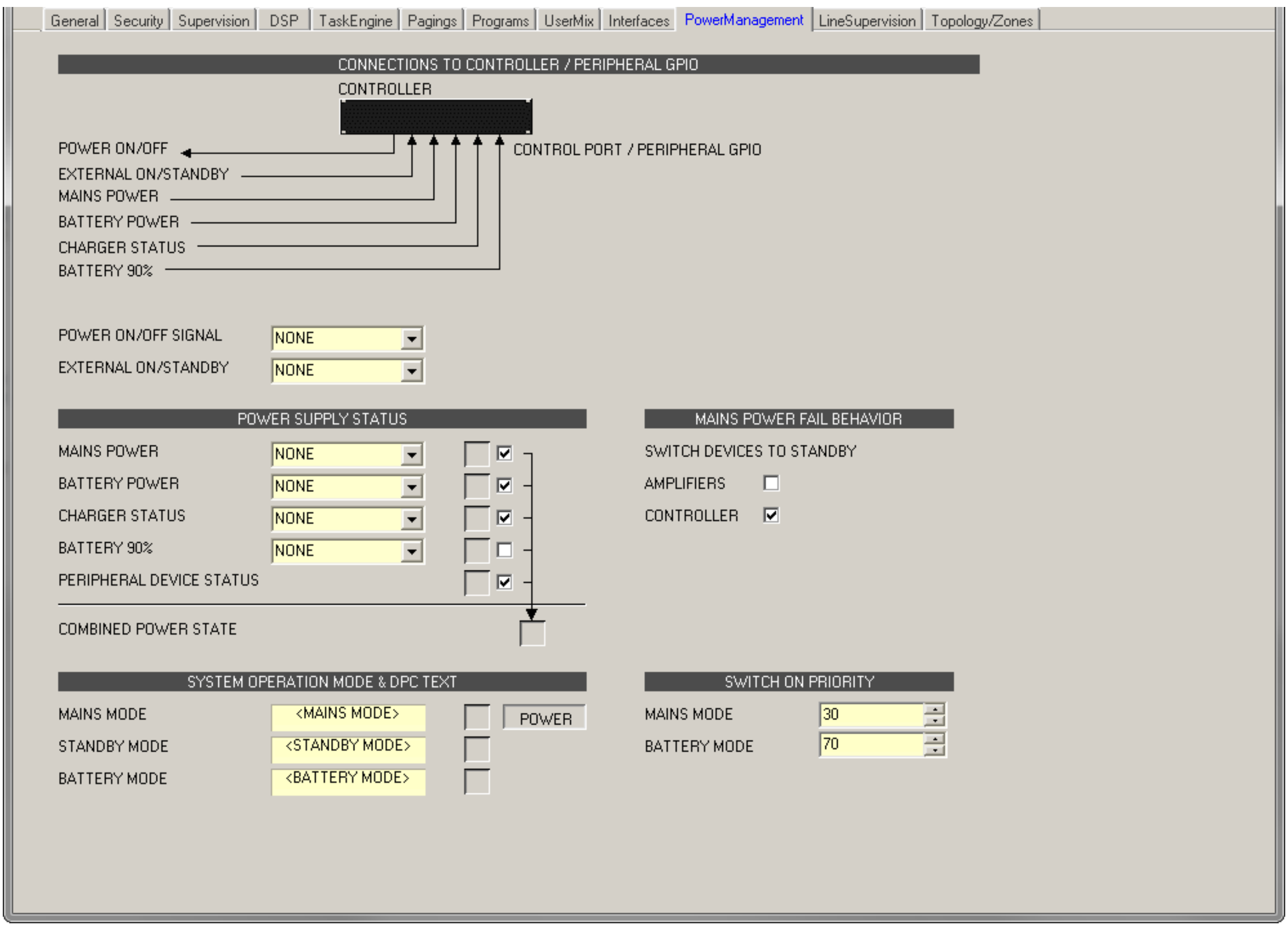
NUMBER OF LOGIC VALUES	Enter the number of logic values of the task engine to be available via the ASCII control protocol.
NUMBER OF ANALOG VALUES	Enter the number of analog values of the task engine to be available via the ASCII control protocol.
IS ALIVE PERIOD (s)	Enter the is alive period of the ASCII control protocol in seconds.
DPM 4000 INTERFACE	Please see section below for details about purpose and configuration.
ENABLE	Set the checkbox to activate the RS-232 interface between a DPM 4000 and the DPM 8016.
AUDIO OUTPUT	Select the audio output of the DPM 8016, that should output the audio signal to the DPM 4000.
AUDIO INPUT	Select the audio input number of the DPM 4000.
MEMOFLAG BROKEN LINE	The DPM 4000 memo flag selected here is used for supervising the RS-232 connection.
CONTROL PORT	
	Clicking with the right mouse button on the corresponding symbol of a control input provides the configuration dialog of this control input. (not yet activated)
	Displays the control inputs' current condition.
	It is possible to manually change the condition of the control outputs (normally open contact / normally closed contact). Depending on their according configuration, control outputs are only switched as long as the mouse button is pressed.
	Clicking with the right mouse button on the corresponding symbol of a control output provides opens the configuration dialog of this control output. (not yet activated)

DPM 4000 Interface
The DPM 4000 interface of the DPM 8016 gives the possibility of extending an existing Promatrix 4000 system by additional zones / call stations / amplifier power, etc.
Following hardware is required:
– a DPM 8016 controller with modules
– at least one DPC call station
– at least one DPA amplifiers
DCS 400 relay cards are optional. The DPC 8015 call station enables to do announcements into the existing Promatrix 4000 zones and new Promatrix 8000 zones. Background music (BGM) assignment is also possible from the DPC 8015. The Promatrix 4000 system may be seen as time master for the complete system. Control communication between the two systems is done via the RS-232 interface. Audio transmission is done from an AO-1 audio output module of the DPM 8016 to a Mic/Line input of the DPM 4000.
Configuration
The configuration of the DPM 4000 interface is done in the corresponding section of the DPM 8016 Interfaces dialog. When selecting the checkbox “ENABLE” a specific topology with 100 zones is created automatically. Please take care to start with this topology, as then the first 100 zones are reserved for the Promatrix 4000 system. Existing zones may be deleted.

Open the “Topology / Zones” dialog and add a non-DPA amplifier, select the DPM Audio output to be used for the PM4000. Connect the non-DPA amplifier with the PM4000 topology. Go back to “Interfaces” dialog and check correct DPM8016 audio output (comboBox “AUDIO OUTPUT”). Choose an audio input at the PM4000 (selection field “AUDIO INPUT”). Please ensure that the PM 4000 system offers a free input (re-programming may be necessary).
For fault indication (fault interface), a Memo Flag can be programmed in the PM 4000 system which is triggered via the PM 8000 by setting the “MEMOFLAG BROKEN LINE” selection field accordingly.
If the PM 4000 time shall be used as system time, the PM 8000 system can act as time slave. In this case, at the DPM 8016 dialog “General”, the time synchronization has to be set to “DPM 4000 synchronisation”.

Power Management dialog

The Power Management dialog allows configuring the standby mode of the DPM 8016 in detail.



Element	Description
POWER ON/OFF SIGNAL	Select the GPO contact or the virtual TE value for signaling the DPM operating mode. In standby mode the GPO is open.
EXTERNAL ON/STANDBY	Select the digital GPI or virtual TE value to be used for switching to standby mode.
POWER SUPPLY STATUS	

MAINS POWER	Select the digital GPI or virtual TE value that is used for signaling „mains power OK“.
BATTERY POWER	Select the digital GPI or virtual TE value that is used for signaling „battery power OK“.
CHARGER STATUS	Select the digital GPI or virtual TE value that is used for signaling „charger status OK“.
BATTERY 90%	Select the digital GPI or virtual TE value that is used for signaling „battery status at least 90%“.
POWER SUPPLY	This LED is green, if all selected power supply status are OK.
SYSTEM OPERATION MODE & DPC TEXT	
MAINS MODE	If mains power is used for running the system, the DPM is in MAINS MODE and the LED lights green. You can edit the name of this mode in the text field.
STANDBY MODE	If the system is in STANDBY MODE, this LED lights green. You can edit the name of this mode in the text field.
BATTERY MODE	If battery power is used for running the system, the DPM is in BATTERY MODE and the LED lights green. You can edit the name of this mode in the text field.
MAINS POWER FAIL BEHAVIOR	
AMPLIFIERS	Select this option if Amplifiers should switch to standby mode if mains power fails.
CONTROLLER	Select this option if the Controller should switch to standby mode if mains power fails..
SWITCH ON PRIORITY	
MAINS MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is available.
BATTERY MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is not available (battery mode).

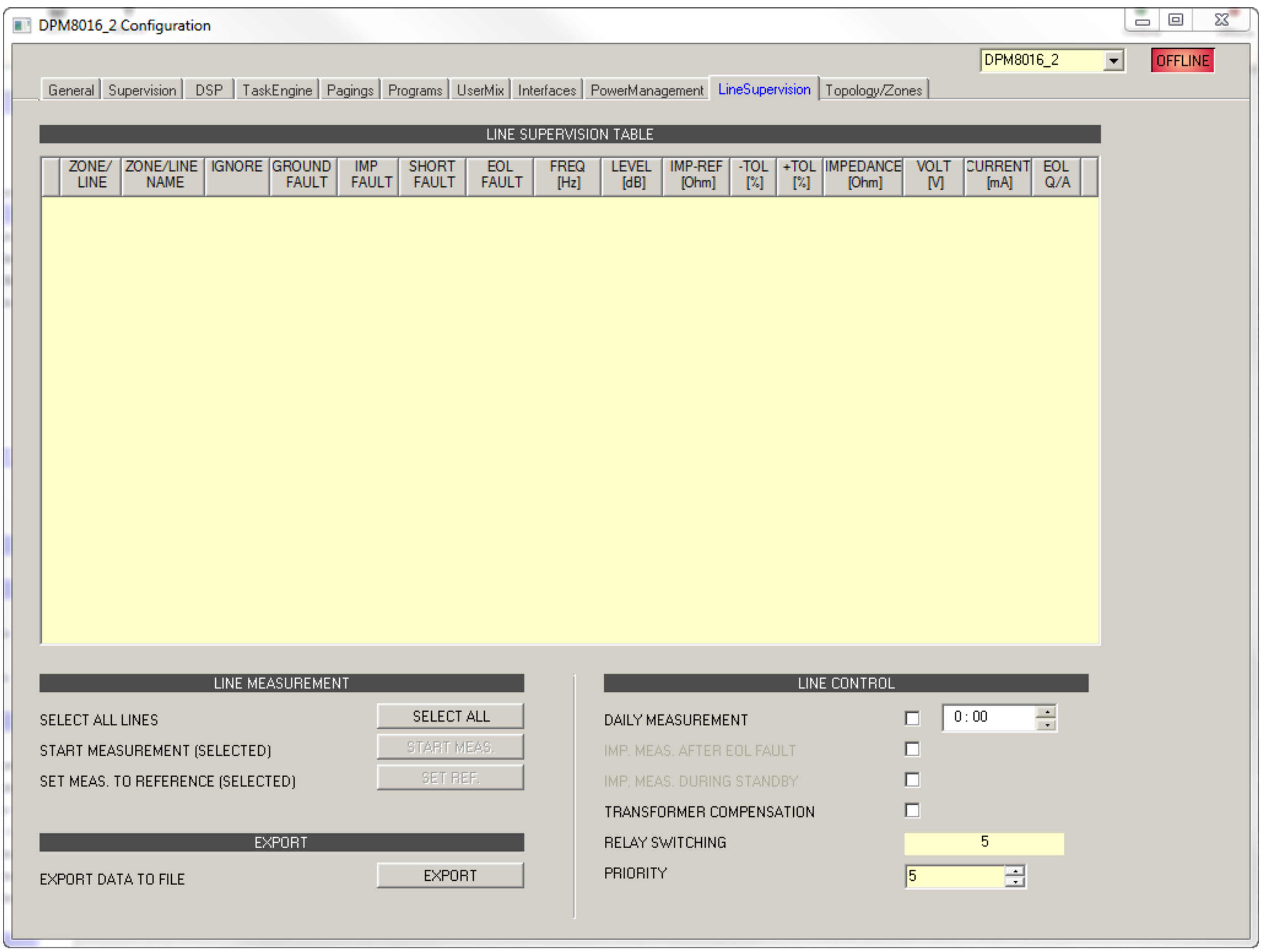
HINT: The “Power Calculator” tool can be used to calculate the power consumption of a DPM 8016 or a complete PROMATRIX 8000 system. The tool can be found in directory “/Tools” or can be requested from the IRIS-Net support team.

HINT: The properties “Operating Mode” and “Standby LED” can be used for advanced power management configuration via the Task Engine, please refer to section *Properties, page 777*.

LineSupervision dialog

The Line Supervision dialog allows configuring and control of the DPM line supervision. Line supervision can be done via line impedance measurement method or the End of Line (EOL) method.

- When using the impedance measurement method the impedance of the speaker line is calculated using a voltage and current measurement. A fault message is indicated in the event that the measurement exceeds or falls short of the tolerance range. The impedance measurement method is not suitable for permanent line supervision. The upper and lower bounds of the impedance measurement, e.g. frequency or impedance value, depend on the amplifier type used, please refer to the amplifier owner‘s manual.
- When using EOL 8001 modules the amplifier is communicating with the modules. A fault message is indicated if the communication fails, e.g. because of a short circuit or wire break. The EOL method allows permanent supervision, acoustic signals do not interrupt or affect the supervision. Usage of EOL 8001 modules is only possible if direct topology is used.



LINE SUPERVISION TABLE

Element	Description
ZONE/LINE	System internal description of the zone or line.

ZONE/LINE NAME	Description of the zone or line.
IGNORE	Check this checkbox, if the result of the line measurement should be ignored. A error of this zone or line will not be indicated in the system. Regular measurements are carried out anyway. **HINT: If the checkbox is checked a short cut will not be indicated. If the zone is connected via a line relay, the relay will be deactivated.**
GROUND FAULT	This LED lights red, if a ground fault error has occurred.
IMP FAULT	This LED lights red, if the measured impedance is out of the tolerance range.
SHORT FAULT	This LED lights red, if there is a short cut at the zone or line (measured impedance value below 25% of reference value). In this case the system will not start calls or alarms in this zone or line. **HINT: If the zone is connected via a line relay, the relay will be deactivated when there is a short cut (short cut protection for other lines on the same amplifier).**
EOL FAULT	This LED lights red, if a EOL error has occurred.
FREQ [Hz]	Enter the frequency of the measurement signal.
LEVEL [dB]	Enter the level of the measurement signal, based on the maximum output level. Example: A setting of -20dB corresponds to a output level of 10 Veff when using a 100 Veff amplifier output.
IMP-REF [Ohm]	Indicates the impedance reference value of the zone or line.
-TOL [%]	Maximum negative deviation of the impedance value of the zone or line from the reference value, given in procent.
+TOL [%]	Maximum positive deviation of the impedance value of the zone or line from the reference value, given in procent.
IMPEDANCE [Ohm]	Indicates the impedance value of the zone or line of the last successful measurement.
VOLT [V]	Indicates the voltage of the measurement signal of the last successful measurement.
CURRENT [mA]	Indicates the current of the measurement signal of the last successful measurement.
EOL Q/A	Indicates the quantity and addresses of the EOL modules in the zone or line.
STATUS	Indicates the status of the measurement.
SELECT ALL	All zone or lines are selected.
START MEAS.	Starts the line measurement in all selected zones or line.
SET REF.	Press this button to store the values of the last measurement as new reference values for the selected zones or lines.
EXPORT	All measurement data of the LINE SUPERVISION TABLE are exported to a csv file. Open the file in a spreadsheet for further processing.

DAILY MEASUREMENT	Check this checkbox, if a daily measurement should be done automatically. Enter the time the measurement should start.
TRANSFORMER COMPENSATION	Check this checkbox to optimize the impedance measurement for high impedance speaker lines (e.g. one speaker only).
RELAY SWITCHING	Enter the number of relay switching cycles, which should be done before the measurement starts. This is only valid for line relays, priority relays, control relays or call relays.
PRIORITY	Priority of the line measurement signal.

The Line Supervision table is automatically generated from the available zones filled with default values.

HINT: Use copy & paste to copy configurations from one element to another element in the line supervision table.

IMPEDANCE METHOD

The values of frequency, level and tolerance can be edited and adapted to the real conditions. To generate the reference values a first line measurement must be performed, the resulting measurement values are stored as reference values. The measurement of the lines and the comparison with the reference values is done automatically every day at the scheduled time if the line is not busy. Each audio signal on the line interrupts the line measurement. The measurements will be continued automatically if the line is free again.

EOL METHOD

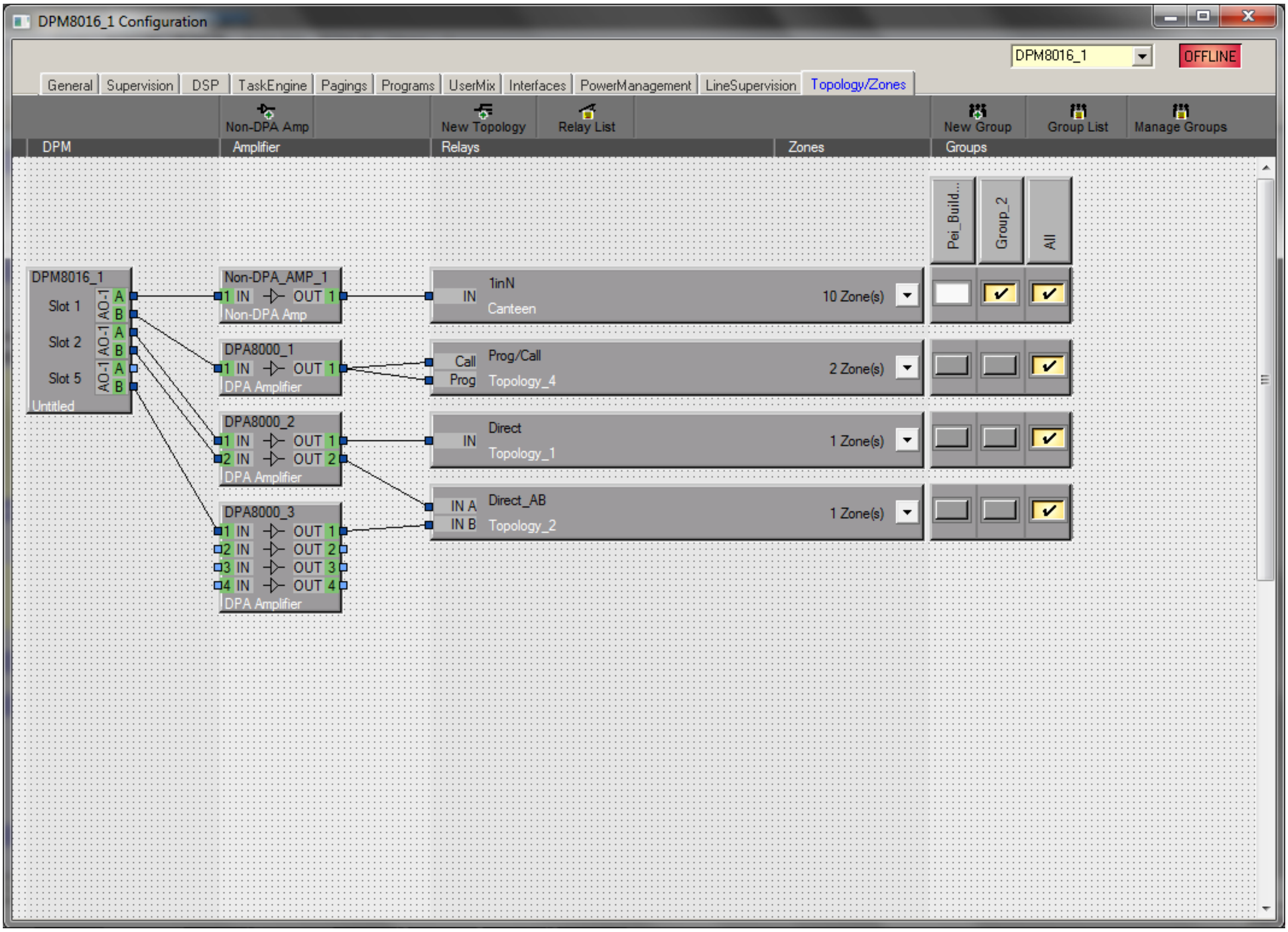
To enable the EOL supervision for a zone or line in the column EOL Q / A in the first line the number of EOL modules connected to the line must be entered, in the following lines the addresses of the modules must be entered. Enter 0 to disables the EOL method for the corresponding line.

HINT: For power supply of the EOL modules a pilot tone is required, so the pilot tone generator of the power amplifier shall be activated.

For more information and technical data for the two measurement methods please refer to the DPA 8000 Owners Manual.

Topology/Zones Dialog

The Topology/Zones dialog window allows configuration of Topologies and Zones. Zones are configured in a topology, each zone can be selected to be member of a Group.

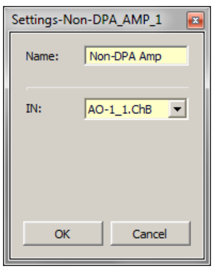


Icon bar

Element	Description
Non-DPA Amp	Click on this button to add a non-DPA amplifier to the Topology/Zone configuration.
New Topology	Click on this button to add a new Topology to Topology/Zones configuration.
Relay List	Click on this button to generate excel sheet report of the configured relays. The report includes relay names, configured zones, type, invert status, topology and relay system object name. Note: If Microsoft Excel is not installed on your computer, a CSV report is generated.
New Group	Click on this button to create a new Group. The All group, including all zones, is created automatically. For every new group the zones can be selected via the checkboxes in the group column.

Element	Description
Group List	Click on this button to generate excel sheet report of all Groups configured in the PROMATRIX 8000 system. The report includes the caption and object id of systems zones and the assignment of zones to system groups. Note: If Microsoft Excel is not installed on computer, a CSV report is generated.
Manage Groups	Click on this button to open the Manage Group Dialog. This dialog allows to add or delete Groups and to add or remove Zones from a selected Group.

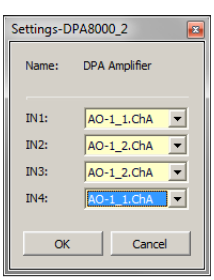
Non-DPA amplifier settings dialog



Element	Description
Name	Name of the amplifier.
IN	Select the input source of the amplifier. This Combo box lists all output channels of AO-1 modules of the DPM 8016.

Notice!
This dialog can be useful to check or edit connections between DPM outputs and amplifier inputs in large systems.

Amplifier settings dialog
This dialog can be opened by doubled clicking on any of the amplifier blocks. This dialog can be used to create or edit connections between output channels of AO-1 modules and amplifier input channel.



Element	Description
Name	Name of the amplifier.
IN1 to IN4	Select the input source of the amplifier channel. This Combo box lists all output channels of AO-1 modules of the DPM 8016. There is a combo box for each channel of the amplifier (1, 2 or 4).

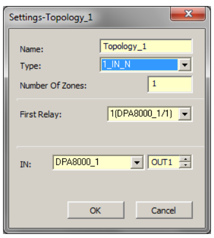
Topologies

This dialog allows adding new topologies to the Topology/Zones configuration. Following four topology types are available:

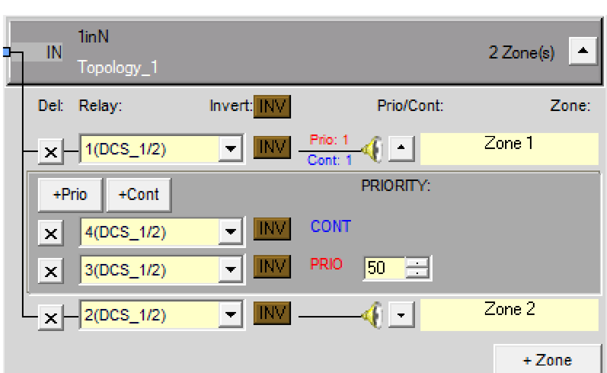
– 1 in N
– Direct
– Direct AB
– Prog/Call

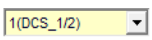
Following tables describe the elements of the four topology setting dialogs and maximized topology blocks.

1 IN N topology

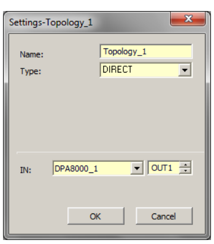


Element	Description
Name	Name of topology
Type	This combo box lists all type of topologies that can be created in the Promatrix 8000 system.
Number Of Zones	Number of zones to be created in the selected topology.
First relay	Number of the first relay to be assigned to first zone created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay. Note: The Relay Name is given as “Relay No (Device_x/Channel or Slot)” with x as number of the device. An example for making this numbering clearer is listed below. If No Relay is selected as first relay, zones are created without assigning any physical device relays to them. In this case the relays have to be assigned manually.
IN	Select the output channel of the amplifier to be used as input for this topology.

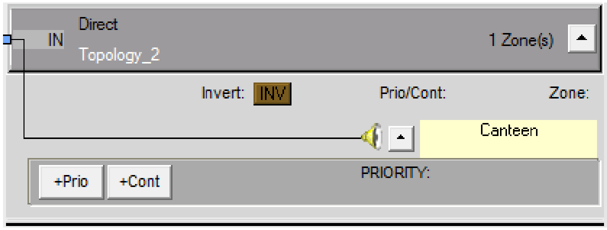


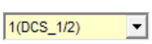
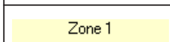
Element	Description
INV	Press the INV button to invert the status of all relays in the topology.
	Press this button to minimize or maximize the zones or relays dialog.
+Prio	Press this button to add a priority relays to the zone. Note: Up to 5 priority relays can be configured in a zone.
+Cont	Press this button to add a control relays to the zone. Note: Up to 5 control relays can be configured in a zone.
X	Press this button to delete the corresponding zone or relay.
 1(DCS_1/2)	This combo box lists all available relays of DCS or DPA devices in the PROMATRIX 8000 system.
Zone 1	Enter a name for the zone.
 50	This control allows setting the priority value of a priority relay.
+Zone	Click on this button to add a new zone to the topology.

DIRECT topology

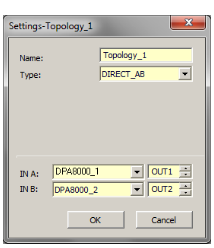


Element	Description
Name	Name of topology
Type	This combo box lists all type of topologies that can be created in the Promatrix 8000 system.
Number Of Zones	Number of zones to be created in the selected topology.
First relay	Number of the first relay to be assigned to first zone created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay. Note: The Relay Name is given as “Relay No (Device_x/Channel or Slot)” with x as number of the device. An example for making this numbering clearer is listed below. If No Relay is selected as first relay, zones are created without assigning any physical device relays to them. In this case the relays have to be assigned manually.
IN A or B	Select the output channel of the amplifier to be used as input for part A or B of this topology



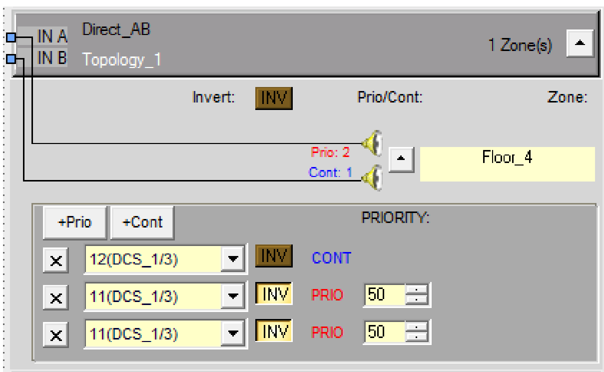
Element	Description
INV	Press the INV button to invert the status of all relays in the topology.
	Press this button to minimize or maximize the zones or relays dialog.
+Prio	Press this button to add a priority relays to the zone. Note: Up to 5 priority relays can be configured in a zone.
+Cont	Press this button to add a control relays to the zone. Note: Up to 5 control relays can be configured in a zone.
X	Press this button to delete the corresponding zone or relay.
	This combo box lists all available relays of DCS or DPA devices in the PROMATRIX 8000 system.
	Enter a name for the zone.
	This control allows setting the priority value of a priority relay.

DIRECT AB



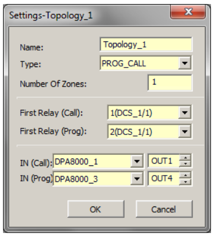
Element	Description
Name	Name of topology
Type	This combo box lists all type of topologies that can be created in the Promatrix 8000 system.
Number Of Zones	Number of zones to be created in the selected topology.

Element	Description
First relay	Number of the first relay to be assigned to first zone created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay. Note: The Relay Name is given as “Relay No (Device_x/Channel or Slot)” with x as number of the device. An example for making this numbering clearer is listed below. If No Relay is selected as first relay, zones are created without assigning any physical device relays to them. In this case the relays have to be assigned manually.
IN A or B	Select the output channel of the amplifier to be used as input for part A or B of this topology

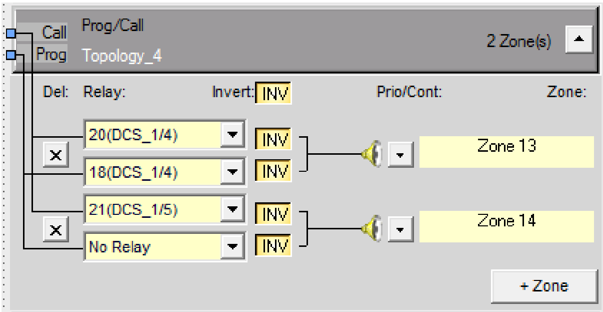


Element	Description
INV	Press the INV button to invert the status of all relays in the topology.
▲	Press this button to minimize or maximize the zones or relays dialog.
+Prio	Press this button to add a priority relays to the zone. Note: Up to 5 priority relays can be configured in a zone.
+Cont	Press this button to add a control relays to the zone. Note: Up to 5 control relays can be configured in a zone.
X	Press this button to delete the corresponding zone or relay.
1(DCS_1/2)	This combo box lists all available relays of DCS or DPA devices in the PROMATRIX 8000 system.
Zone 1	Enter a name for the zone.
50	This control allows setting the priority value of a priority relay.

PROG CALL



Element	Description
Name	Name of topology
Type	This combo box lists all type of topologies that can be created in the Promatrix 8000 system.
Number Of Zones	Number of zones to be created in the selected topology.
First relay	Number of the first relay to be assigned to first zone created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay. Note: The Relay Name is given as “Relay No (Device_x/Channel or Slot)” with x as number of the device. An example for making this numbering clearer is listed below. If No Relay is selected as first relay, zones are created without assigning any physical device relays to them. In this case the relays have to be assigned manually.
First Relay (Call)	Number of the first relay to be assigned to first call line created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay.
First Relay (Prog)	Number of the first relay to be assigned to first program line created in this topology. Relays are automatically assigned to the subsequent Zones in ascending order starting from the first relay.
IN (Call)	Select the output channel of the amplifier to be used as input for the call line of this topology
IN (Prog)	Select the output channel of the amplifier to be used as input for the program line of this topology



INV	Press the INV button to invert the status of all relays in the topology.
	Press this button to minimize or maximize the zones or relays dialog.
+Prio	Press this button to add a priority relays to the zone. Note: Up to 5 priority relays can be configured in a zone.
+Cont	Press this button to add a control relays to the zone. Note: Up to 5 control relays can be configured in a zone.
X	Press this button to delete the corresponding zone or relay.
1(DCS_1/2)	This combo box lists all available relays of DCS or DPA devices in the PROMATRIX 8000 system.
Zone 1	Enter a name for the zone.
50	This control allows setting the priority value of a priority relay.
+Zone	Click on this button to add a new zone to the topology.

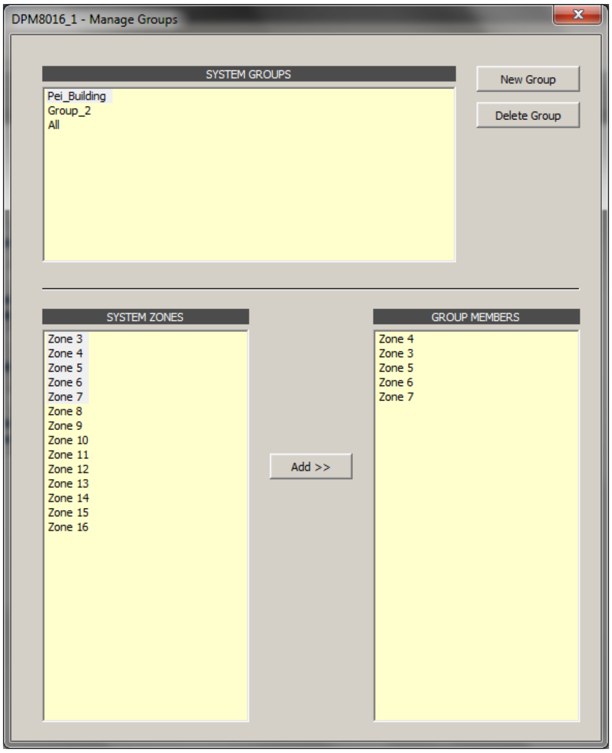
Relay numbering

Following table lists the names given to relays of a two channel DPA 8225 amplifier.

Relais of amplifier	Internal name used in topology settings dialog
Relay 1 of channel 1	1(DPA8000_1/1)
Relay 2 of channel 1	2(DPA8000_1/1)
Relay 1 of channel 2	3(DPA8000_1/2)
Relay 1 of channel 2	4(DPA8000_1/2)

Manage group dialog

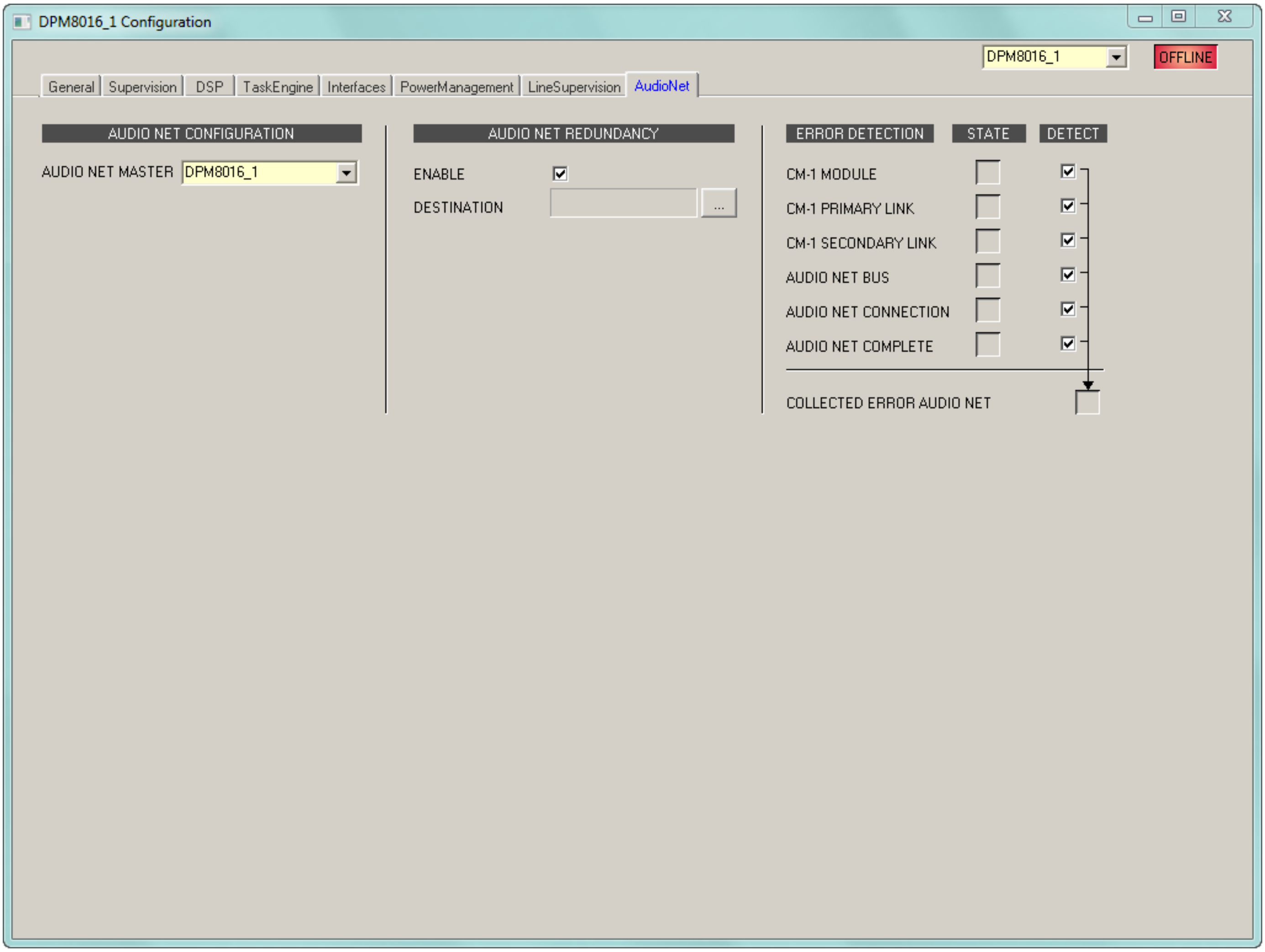
This dialog allows to create, edit or delete groups. It is also possible to add or remove zones from a selected Group. To remove a zone from a group, select the zone in the GROUP MEMBERS section and press the delete button.



Element	Descripiton
SYSTEM GROUPS	Lists all groups of the PROMATRIX 8000 system.
New Group	Press this button to create a new group.
Delete Group	Press this button to delete the group selected in the SYSTEM GROUPS list.
SYSTEM ZONES	Lists all zones of the PROMATRIX 8000 system.
Add >>	Adds the zones selected in the SYSTEM ZONES list to the group selected in the SYSTEM GROUPS list.
GROUP MEMBERS	Lists the zones currently included in the group selected in the SYSTEM GROUPS list.

AudioNet Dialog

The AudioNet dialog allows the configuration and supervision of an audio network, consisting of two or more DPM 8016 connected via CM-1 CobraNet modules. The AudioNet tab will only be displayed when a CM-1 CobraNet module is configured. With AudioNet it is possible to transmit audio signals from one DPM to one or more other DPM. One DPM needs to be configured as AudioNet master.
All other DPM within the project are automatically configured as AudioNet slave. Both CobraNet interface (audio) and Ethernet (control data) are required for setting up a AudioNet. Upon failure of the primary or the secondary CobraNet interface the other interface will be activated automatically. In case of failure of the Ethernet AudioNet Redundancy is enabled. This function allows transmitting a signal in a predefined zone pattern to all DPMs, if the function is activated and a Destination is defined.



Element	Description
AUDIO NET MASTER	Select the DPM 8016 from the Dropdown menu that should be used as master in the audio network. **HINT: In large networks usage of a network metric, e.g. betweenness centrality is recommended for selecting the master**.
ENABLE	Check this checkbox, if a redundant audio network is used.
DESTINATION	Click the ... button to open the Destinations dialog. The Destinations dialog allows selecting a zone/group. If there is an error in the audio network, the audio signal is transmitted to the selected zone/group. **HINT: Selecting the “all” group is recommended.**
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR AUDIO NET flag is set at the same time.
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
CM-1	Error in CM-1 module of the DPM 8016.

CM-1 PRIMARY LINK	Error in the connection of the PRIMARY LINK interface to another network device (e.g. switch, router, DPM).
CM-1 SECONDARY LINK	Error in the connection of the SECONDARY LINK interface to another network device (e.g. switch, router, DPM).
AUDIO NET BUS	Error in audio network (CobraNet).
AUDIO NET CONNECTION	Error in connection to audio network (Slave DPM has no Ethernet connection to Master DPM).
AUDIO NET COMPLETE	The number of devices connected to the audio network (Ethernet) is not identical to the number of configured devices.
COLLECTED ERROR AUDIO NET	This error is indicated as “AUDIO NET” error in the Supervision dialog.

Properties

BUZZER CONTROL

The DPM8016_1.BuzzerControl property of the DPM 8016 allows configuring the integrated buzzer. Following settings are available:

Value	Description
on	Buzzer is activated is a new error appears.
off	Buzzer is deactivated.
DPC_1	Buzzer is activated if the call station (DPC_1, DPC2,...) is not connected.

OPERATION MODE

The “DPM8016_1.System.PowerManagement.OperatingMode” property allows setting the current operation mode of the DPM 8016 and connected devices. High priority signals prevent changing into standby mode. Following settings are available:

Value	Description
0	Switch DPM 8016 in standby mode
1	Switch DPM 8016 in operating mode

HINT: The mode of peripheral devices connected to the DPM 8016 is set automatically.

STANDBYLED

The Standby LED of the DPM 8016 lights, when the device is in standby mode. The corresponding property “DPM8016_1.System.StandbyLED” can be used to query the current mode.

Value	Description
0	DPM 8016 is in operating mode
1	DPM 8016 is in standby mode

ASCII Control Protocol

The DPM 8016 (firmware version V1.8.0 or higher) can be easily integrated into media or touch panel controls via the Ethernet interface. This section describes how to connect via Ethernet and the available options of the control protocol.

ETHERNET CONNECTION

To connect a external device to the DPM 8016 via Ethernet, the Ethernet port of the DPM must be configured. Following table lists the keywords of the DPM for Ethernet configuration.

Keyword	Values	Default	Description
OpenIntActive	0, 1	0	This keyword allows activating or deactivating the ASCII Control Protocol. If deactived, a connection via Ethernet is not possible. – OpenIntActive = 1: The ASCII Control Procol is activated – OpenIntActive = 0: The ASCII Control Procol is not activated
OpenIntPort		6273	The Ethernet port for the TCP connection between the DPM and the external device.

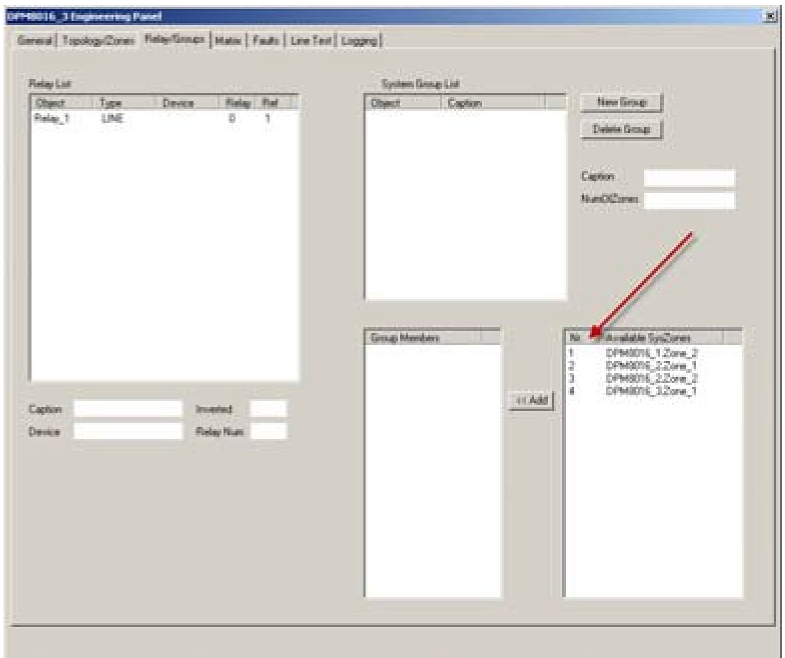
Following table lists the default setting of the DPM 8016 Ethernet port.

Parameter	Default
IP adress	192.168.1.100
Network mask	255.255.255.0
Standard gateway	192.168.1.1
Port	6273

HINT: Only one single Ethernet connection can be used at one time.
HINT: The password protection of the ASCII Control Protocol can be configured at the Interfaces dialog of the DPM 8016, see page 542.

ZONE STATUS
Whenever the output status of the local zones changes, a zone status string is sent via the ASCII Control Protocol. The format of the zone status string is: <Idx.y> <Change> <Pm> <Zone Pattern>

Element	Description
Idx.y	Unique identifier: x is the number x in „DPM8016_x“ in IRIS-Net y is a unique paging request number of the local DPM
Change	ON, if the paging request Idx.y has activated one or more local audio outputs of the DPM OFF, if the paging request was terminated and the corresponding local audio outputs of the DPM have been deactivated

Pm	Priority of the paging request
Zone Pattern	List of activated local zones of the DPM. **HINT: The system zone numbers are listed, please refer to the Relays/Groups tab of the DPM8016 Engineering panel.** **Example:** In the example shown in following figure zone 2 of DPM8016 1 has number 1, the zones 1 and 2 of DPM 8016_2 have the numbers 2 and 3, the zone 1 of DPM8016 3 has number 4. 

Examples:
Id3.7 ON P12 Z2-3
Id3.7 OFF

WATCHDOG FUNCTION
If the ASCII Control Protocol is activated and also the watchdog string output is activated, following string is sent via the ASCII Control Protocol at regular intervals: “Open IntIs Alive”

Keyword	Range	Default	Description
Open IntIs Alive Period	0, 1,..., 100	0	This keyword is used to active or deactivate the output of the watchdog string. When activated the output time period can be adjusted. – Open IntIs Alive Period = 0: No output of the watchdog string. – Open IntIs Alive Period = 1...100: The watchdog string is output every 1...100 seconds. **Example:** Open IntIs Alive Period = 15: The watchdog string is output every 15 seconds.

INPUT VIA THE ASCII CONTROL PROTOCOL

The ASCII Control Protocol allows entering values for input parameters (rational numbers or Boolean values) of the DPM Task Engine. Following keywords must be used to setup variables in IRIS-Net.

Keyword	Range	Default	Description
OpenIntLValNrof		10	Sets the number of logic values (Boolean values)
OpenIntAValNrof		10	Sets the number of analog values (rational numbers)

Logic values

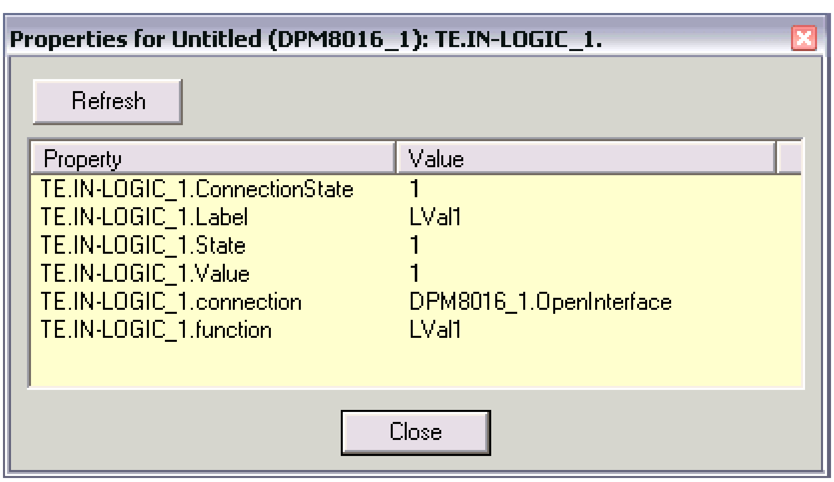
Following format is used to edit the value of a logic value: <LVal><id> <value>

Element	Description
LVal	Indicates the use of a logic value in the Task Engine
id	Unique identifier of the logic value in the Task Engine
value	Boolean value to be assigned to the logic value

Example:

LVal1 0

For assigning a Boolean value via the ASCII Control Protocol the Task Engine block type „Input Logic“ is used. The value of the block‘s property function must be set to <LVal><id>. The value of the block‘s property connection must be set to DPM8016_x.OpenInterface



Analog values

Following format is used to edit the value of a analog value: <AVal><id> <value>

Element	Description
AVal	Indicates the use of a analog value in the Task Engine
id	Unique identifier of the analog value in the Task Engine
value	Rational number to be assigned to the analog value

Example:
AVal7 -30.222
For assigning a rational number via the ASCII Control Protocol the Task Engine block type „Input Analog“ is used. The value of the block‘s property function must be set to <AVal><id>.

OUTPUT VIA THE ASCII CONTROL PROTOCOL
The ASCII Control Protocol allows to query values (rational numbers or Boolean values) of the DPM Task Engine.
Logic values
Following format is used to query the value of a logic value: <LVal><id> ?

Element	Description
LVal	Indicates the use of a logic value in the Task Engine
id	Unique identifier of the logic value in the Task Engine
?	Question mark

Example:
Query: “LVal 7?”
Reponse: “LVal7 1”
Analog values
Following format is used to query the value of a analog value: <AVal><id> ?

Element	Description
AVal	Indicates the use of a analog value in the Task Engine
id	Unique identifier of the analog value in the Task Engine
?	Question mark

Example:
Query: “AVal 7”
Response: “AVal7 -30.2222”

Caution!
Query-response sequences are not synchronized. E.g. zone status messages could be output between query and response, see following example.
Consequences

Example of unsynchronized output:
AVal 7?
Id3.4 ON P12 Z3,Z5-12,Z15 AVal7 -30.2222

FAULT REPORTING
The ASCII Control Protocol can be used to report faults to external systems. The set of error types to be output via the ASCII Control Protocol can be configured at the Supervision dialog of the DPM 8016, (see *Supervision Dialog, page 718*)
Following format is used to report faults: Fault <Id>#<Parameter> <State> “<DPCText>”

Element	Description
Id	Stable fault number of fault IDs, see table below.
Parameter	Stable fault parameter of error value, see table below.
State	0 if fault disappeared, 1 if fault occurred
DPC Text	User defined error message including the variable %u substituted with a parameter value

Examples:
Module fault on module 2 occurred: “Fault INT-4#2 1” “Module Fehler: #2” Temperature fault disappeared: “Fault INT-6#1 0” “Temperature Fehler”
Following table lists the system error types and corresponding fault IDs of the ASCII control protocol.

GROUP	SYSTEM		OPEN INTERFACE			TASK ENGINE AND LOGGING FAULT GROUPS
	ERRORTYPE	PARAMETER	FAULT ID	PARAMETER	FAULT OUTPUT	
INTERNAL	MEMORY / DATA		INT-1		Fault INT-1#1 1 „DPM Data fault“	Fault group 17:1 asserted
	WATCHDOG		INT-2		Fault INT-2#1 1 „Watchdog fault“	Fault group 3 asserted
	FIRMWARE		INT-3		Fault INT-3#1 1 „DPM software fault“	Fault group 18 asserted
	MODULES	Slot nr. of faulty module in ascending order	INT-4	Slot nr. of faulty module in ascending order	Fault INT-4#2 1 „Module fault:#2“	Fault group 5:2 asserted
	HARDWARE		INT-5		Fault INT-5#1 1 „DPM hardware fault“	Fault group 4 asserted
	TEMPERATUR E		INT-6		Fault INT-6#1 1 „Temperature fault“	Fault group 2 asserted
	AUDIO PROCESSING		INT-7		Fault INT-7#1 1 „DSP system fault“	Fault group 7 asserted
	INPUT PILOT DETEC- TION	Nr. of faulty input channel in ascending order	INT-8	Nr. of faulty input channel in ascending order	Fault INT-8#2 1 „Audio In fault:#2“	Fault group 16:2 asserted
INTERFACES	CAN BUS		IF-1		Fault IF-1#1 1 „Can bus fault“	Fault group 8 asserted

	PCA BUS	Slot nr. of faulty module in ascending order	IF-2	Slot nr. of faulty module in ascending order	Fault IF-2#2 1 „PCA bus fault:#2“	Fault group 9 asserted
	AUDIO NET		IF-3		Fault IF-3#1 1 „AudioNet fault“	Fault group 10 asserted
EXTERNAL	DPC DEVICES	Address of faulty DPC	EXT-1	Address of faulty DPC	Fault EXT-1#2 1 „DPC fault:#2“	Fault group 13:2 asserted
	DPA DEVICES	Address of faulty DPA	EXT-2	Address of faulty DPA	Fault EXT-2#2 1 „DPA fault:#2“	Fault group 12:2 asserted
	DCS DEVICES	Address of faulty DCS	EXT-3	Address of faulty DCS	Fault EXT-3#2 1 „DCS fault:#2“	Fault group 11:2 asserted
	POWER SUPPLY		EXT-4		Fault EXT-4#1 1 „Power fault“	Fault group 19 asserted
	SPEAKER LINE FAULT	Line-Nr. of faulty line with 1 to 500: Zone A, 501 to 1000: Zone B	EXT-5	Line-Nr. of faulty line	Fault EXT-5#100 1 „Line fault:#100“	Fault group 16:100 asserted
USER	USER FAULT 1	User defined	USR-1	User defined	Fault USR-1#17 1 „User fault 1:#17“	Fault group 20:17 asserted
	USER FAULT 2	User defined	USR-2	User defined	Fault USR-2#17 1 „User fault 2:#17“	Fault group 21:17 asserted
	USER FAULT 3	User defined	USR-3	User defined	Fault USR-3#17 1 „User fault 3:#17“	Fault group 22:17 asserted
	USER FAULT 4	User defined	USR-4	User defined	Fault USR-4#17 1 „User fault 4:#17“	Fault group 23:17 asserted
	USER FAULT 5	User defined	USR-5	User defined	Fault USR-5#17 1 „User fault 5:#17“	Fault group 24:17 asserted
	USER FAULT 6	User defined	USR-6	User defined	Fault USR-6#17 1 „User fault 6:#17“	Fault group 25:17 asserted
	USER FAULT 7	User defined	USR-7	User defined	Fault USR-7#17 1 „User fault 7:#17“	Fault group 26:17 asserted
	USER FAULT 8	User defined	USS-8	User defined	Fault USR-8#17 1 „User fault 8:#17“	Fault group 27:17 asserted

	USER FAULT 9	User defined	USR-9	User defined	Fault USR-9#17 1 “User fault 9:#17”	Fault group 28:17 asserted
	USER FAULT 10	User defined	USR-10	User defined	Fault USR-10#17 1 “User fault 10:#17”	Fault group 29:17 asserted

DPC 8000 Call Station

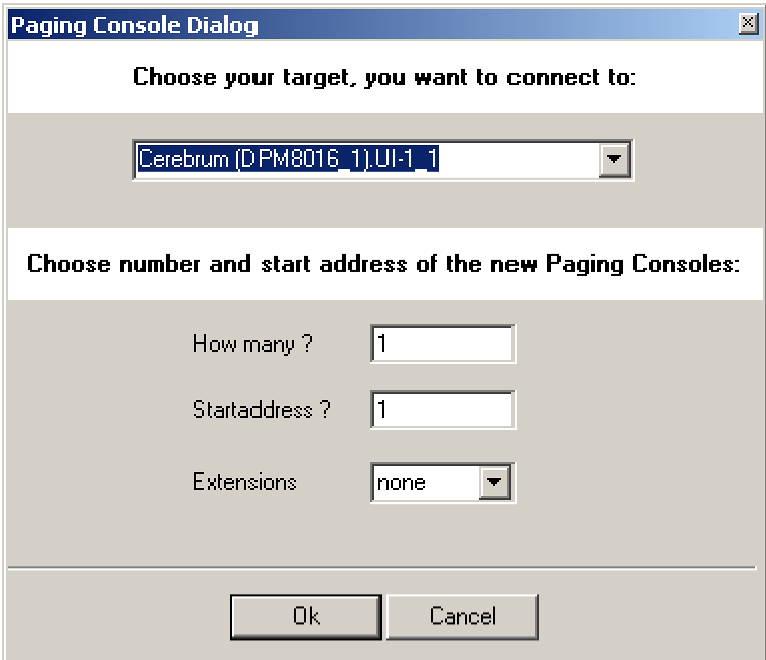
The DPC 8015 is a call station for the PROMATRIX 8000 system. The call station employs a permanently supervised gooseneck microphone with windscreen, a total of 20 buttons, a lighted LC-display and an integrated loudspeaker. The 15 function keys can be permanently assigned for discrete zone addressing. The call station can be expanded to specific requirements with up to five DPC 8120 call station extensions with 20 freely programmable function or zone keys per DPC 8120.

Additional DPC 8015 Features:
- Five pre-programmed function keys - Green LED for each button
- 15 freely programmable function/zone keys - Two LEDs (green/yellow) for each button
- Labeling with transparent cover - Easy editing of key labeling with template file
- Usable as desktop version as well as mounted in consoles/racks
- Internal supervision with event log - fulfills all requirements of relevant national and international standards

DPC 8015 Device

DPC 8015 Device

Start by creating an DPC 8016 Device in your IRIS-Net project. Drag a DPC 8015 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

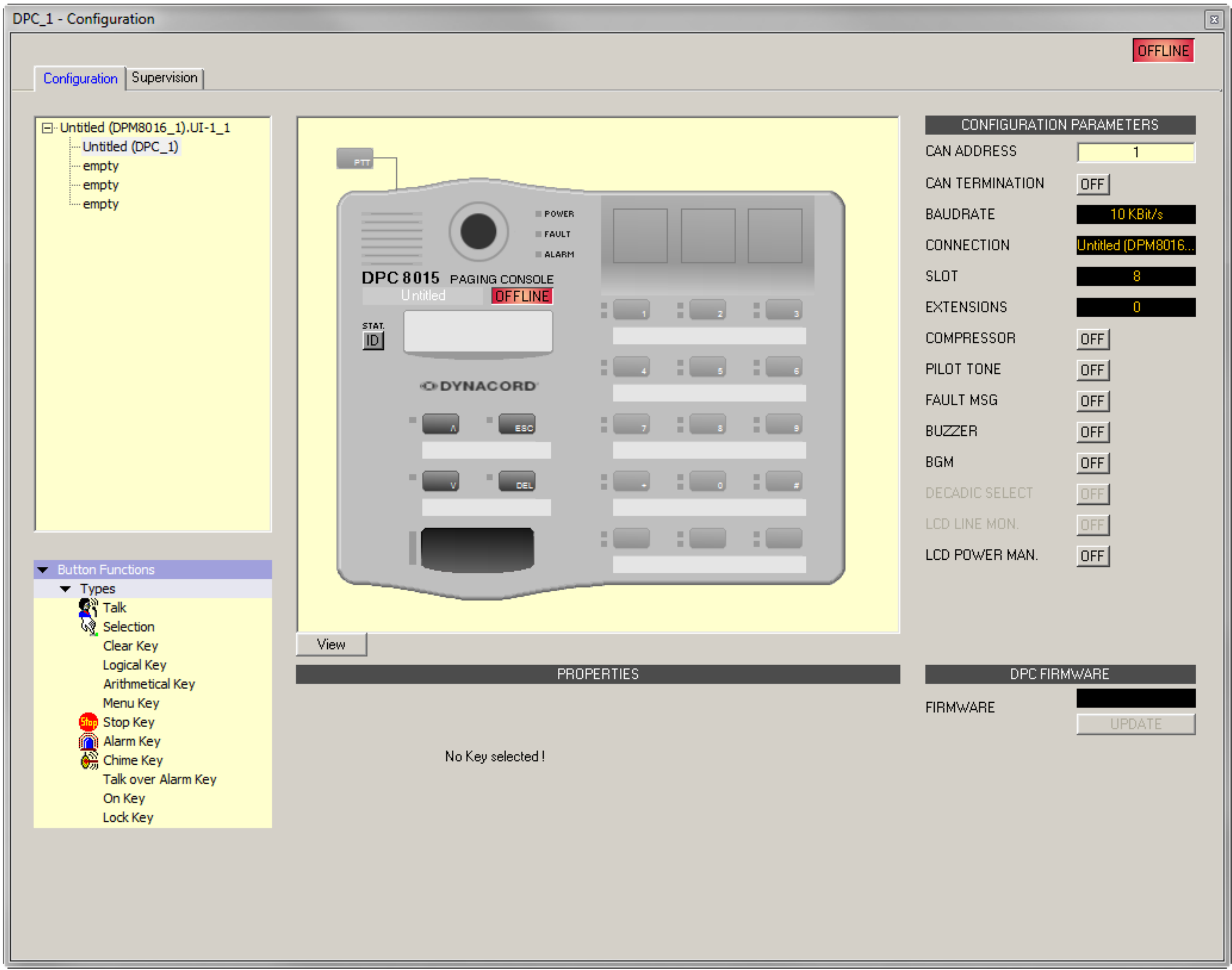


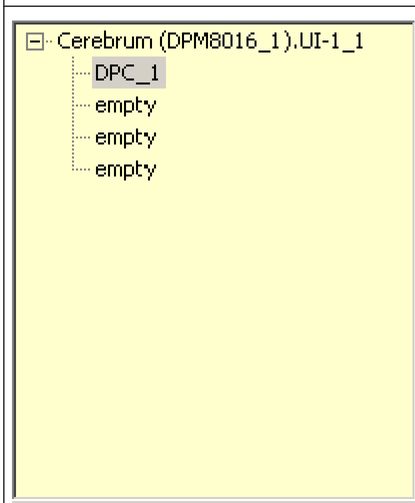
Select the UI-1 Universal Input Module of the DPM 8016 the call station is connected to.
Specify the desired number of devices, the address of the call station and number of call station extensions (it is not possible to add extensions to a call station kit). Click on the OK button to accept these settings.
The specified number of Call Stations will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on a Call Station device icon opens the configuration dialog window. Double clicking on a device for the first time will open the Configuration dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the Call Station device icon.
The following table lists all available DPC 8015 dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

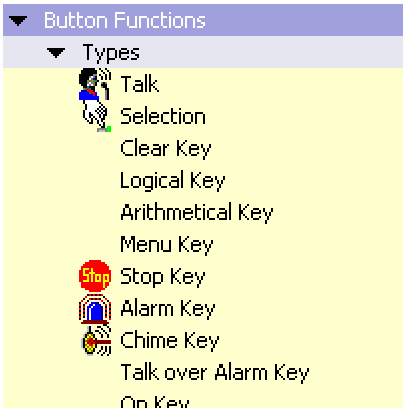
Dialog	Description
Configuration	This window allows hardware settings to be configured, e.g. button configuration, network settings, device name.
Diagnostics	This window provides an overview of the operational state and current fault status of the call station.

Configuration

This page allows making basic settings and retrieve information, for example of button functions, network settings, device name, firmware version, etc.



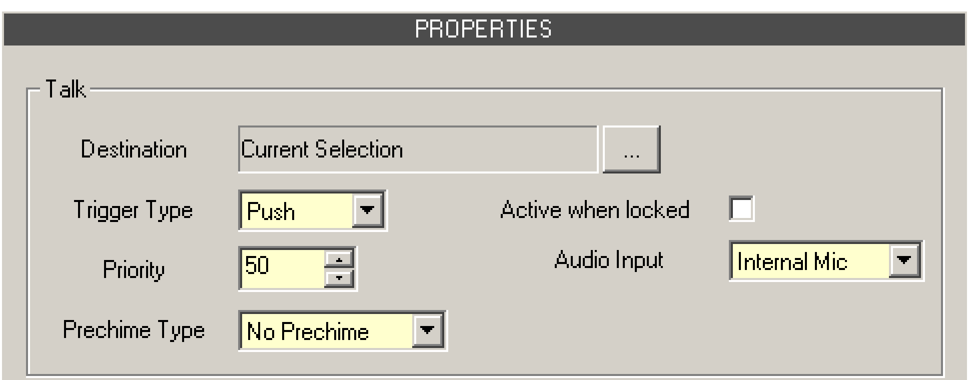
Element	Description
	When there are several DPM 8016 in the project or there are several call stations connected to a single UI-1 Universal Input Module you can select the call station (kit) to configure here.

	Select the desired button type and drag it from this dialog box onto the button of a call station or a call station extension. Detailed information about different types of buttons is provided on the following pages.
 	The Online / Offline indicator signals whether the call station is included in the network or off-line. The red OFFLINE indicator signals that the corresponding call station is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding call station is on-line and that sending and recei- ving data is possible. When on-line, any parameter changes are immediately transmitted and active.
CAN ADDRESS	Displays and lets the user enter the CAN address of the call station. Left-click in the ENTER field and enter the desired address in the range from 1 to 16. The entered value is adopted by pressing RETURN. The entered address has to match the setting in the call station's menu and may only exist once on the CAN-Bus. When adding new call stations to an IRIS-Net project, CAN addresses are automatically assigned in ascending order.
CAN TERMINATION	Press this button (ON) to activate the internal termination resistor of the CAN bus in the call station.
BAUDRATE	The baud rate of the call station. Defining the baud rate is performed via the UI-1 Universal Input Module of the DPM 8016.
CONNECTION	Name of the UI-1 module and the DPM 8016, the call station is connected to.
SLOT	Number of the slot, the UI-1 module is assembled to.
EXTENSION	Number of DPC 8120 extensions.
COMPRESSOR	Press this button (ON) to activate the internal compressor of the call station.
PILOT TONE	Press this button (ON) to activate the pilot tone supervision of the call station. **HINT: When using the pilot tone supervision only one call station can be connected to a PCA bus.**
FAULT MSG	Press this button (ON) if error messages should be indicated in the LC-display of the call station.
BUZZER	Press this button (ON) if errors should be signaled via the integrated buzzer.

BGM	Press this button (ON) if the BGM menu should be accessible in the LC-display of the call station.
	When pressing this button, the backlight of the call station's LCD screen blinks regularly in quick succession. The status indicator of the call station Device in IRIS-Net blinks at the same time. This function serves for checking communication and for identification or search of a call station in a larger system.
	Switching between the following views of a call station and (if existing) call station extensions: – Scroll View – Overall View – Selective View
FIRMWARE	Indicates the firmware version of the DPC when on-line.
UPDATE	Press this button to update the firmware of the call station.

TYPES OF SWITCHES

Talk
A switch of the type "Talk" allows configuring a TALK button. Specific Zones and/or Groups can be pre-selected for this key. Pressing the button on the call station automatically selects the Zones and/or Groups in which the spoken message is being heard.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Trigger (triggers a function)
Priority	Select the button's priority (0 to 9).
Audio Input	Select one of the following audio sources for the announcement: – Internal Mic – External Mic – External Line

Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Prechime Type	Select the desired type of pre-gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – No Prechime – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime

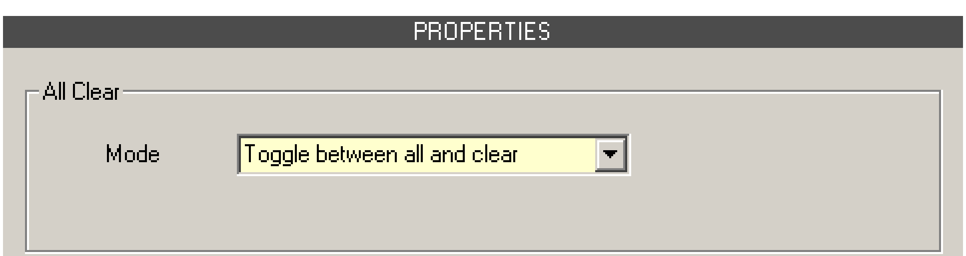
Selection

A switch of the type "Selection" allows configuring a SELECT button. Pressing the button on the call station selects the Zones and/or Groups that have been configured here.

Element	**Description**
Destinatio n	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.

Clear Key

A switch of the type "Clear Key" allows configuring an ALL/CLEAR button. Pressing the button on the call station selects or deselects all Zones and/or Groups.



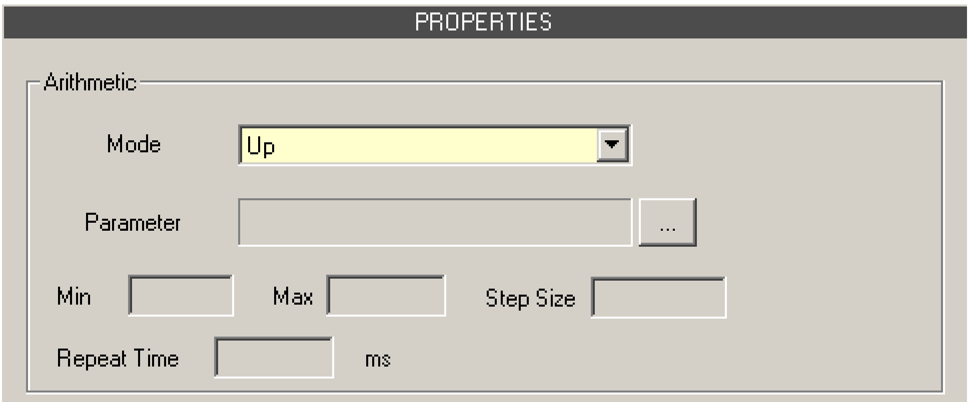
Element	Description
Mode	Select the function that is to be executed when pressing the button on the call station: – Toggle between all and clear = Each press of the button alternately selects or deselects all Zones and/or Groups. – Select All = Pressing the button selects all Zones and/or Groups of the whole system. – Deselect All = Pressing the button deselects all Zones and/or Groups.

Logical Key

A switch of the type "Logical Key" allows setting the value of a logic variable (0 or 1). Pressing the button on the call station sets the logic variable to the desired value. The adjacent LED is operated according to the resulting parameter.

Element	Description
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Set Value = sets the value of the logic variable to "1". It remains "1", even after the button is being released. – Reset Value = sets the value of the logic variable to "0". It remains "0", even after the button is being released. – Push = sets the value of the logic variable to "1", but only as long as the button is being pressed. – Toggle = inverts the value of the logic variable each time the button is being pressed. – LED only = indicates the value of the logic variable, the value is not changed by the button
On	Select the LED of the button that should indicate the value „1“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Off	Select the LED of the button that should indicate the value „0“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Parameter	The logic variable whose value is being changed.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Arithmetical Key
A switch of the type "Arithmetical Key" allows changing the value of a numerical variable. Pressing the button on the call station either increases or decreases the value of the numerical variable.

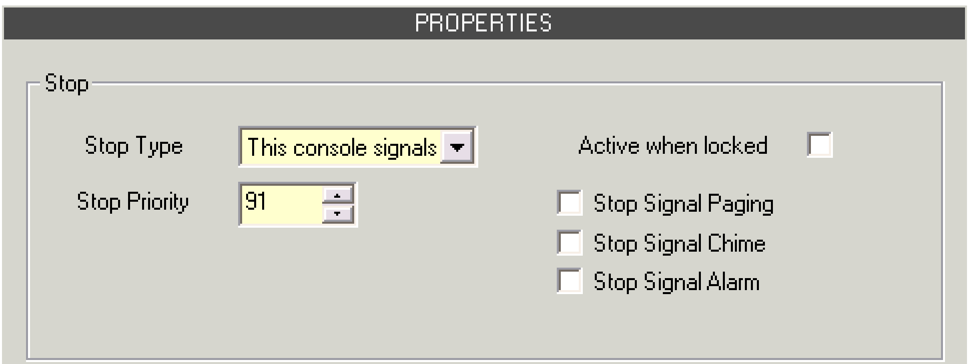


Element	Description
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Up = increases the value of the numerical variable – Down = decreases the value of the numerical variable
Parameter	The numerical variable whose value is being changed.
Min	The lower limit of the value range. Using the "Down" mode decreases the value of the numerical variable till down to this value.
Max	The upper limit of the value range. Using the "Up" mode increases the value of the numerical variable till up to this value.
Step Size	Lets the user enter the step width by which the value is to be changed when pressing the button on the call station.
Repeat Time	Lets the user enter a value for the time interval in milliseconds after which (when keeping the button pressed) the numerical value is being changed by the set step width art any one time. Entering "0" changes the value only once, even when keeping the button pressed over a longer period of time.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Menu Key
A switch of the type "Menu Key" displays the menu on the LCD screen of a call station.

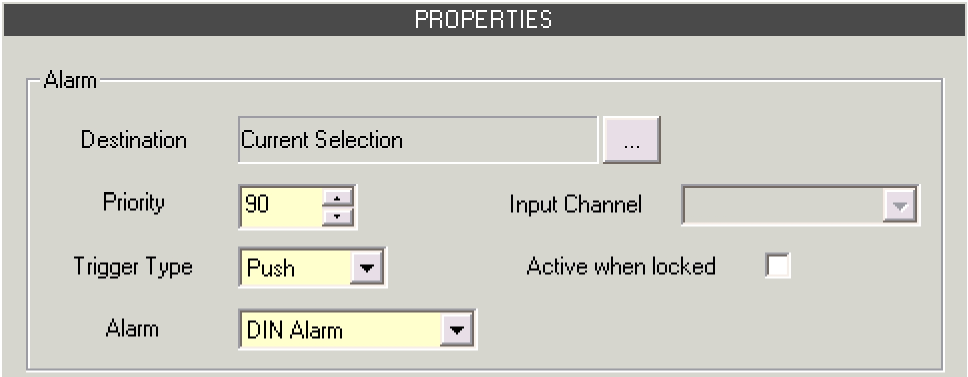
Element	Description
Jump to	Select the position in the menu structure that is to be displayed

Stop Key
A switch of the type "Stop" allows canceling a process that is currently running on the system.



Element	Description
Stop Type	Select the function that is to be executed when pressing the button on the call station: – This concsole signals (local actions) = stops only the types of actions that have been launched from this specific call station – System signals = stops all selected types of actions system-wide, even if they have not been launched from this specific call station
Stop Priority	Select the maximum priority for the signals that will be stopped when pressing the button on the call station.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Stop Signal Paging	Pressing the button on the call station stops pagings.
Stop Signal Chime	Pressing the button on the call station stops chimes.
Stop Signal Alarm	Pressing the button on the call station stops alarms.
Stop Signal Text	Pressing the button on the call station stops signal texts.

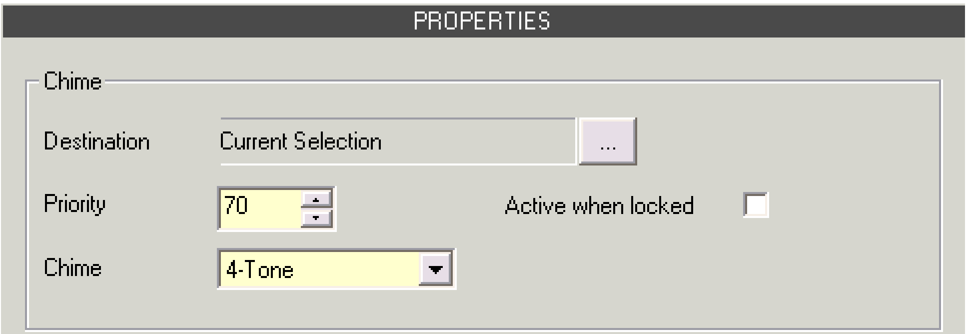
Alarm key
A switch of the type "Alarm" allows starting an Alarm on the system.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the alarm priority (0 to 100).
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger (triggers a function)
Alarm	Select the desired signal that is to be used for alarming: – Extern – DIN Alarm – Slow Whoop – Siren – Two-Tone Alarm – Telephone Alarm – Ship Alarm 1 – Ship Alarm 2 – Ship Alarm 3 – Ship Alarm 4 – Ship Alarm 5 – Ship Alarm 6 – Ship Alarm 7 – Ship Alarm 8 – Ship Alarm 9 – Ship Alarm 10 – Ship Alarm 11 – Ship Alarm 13 – Ship Alarm 14 – Ship Alarm 15 – Ship Alarm 16 – Ship Alarm 17
Input Channel	Enter the audio input at which the externally generated alarm signal is present.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Chime Key

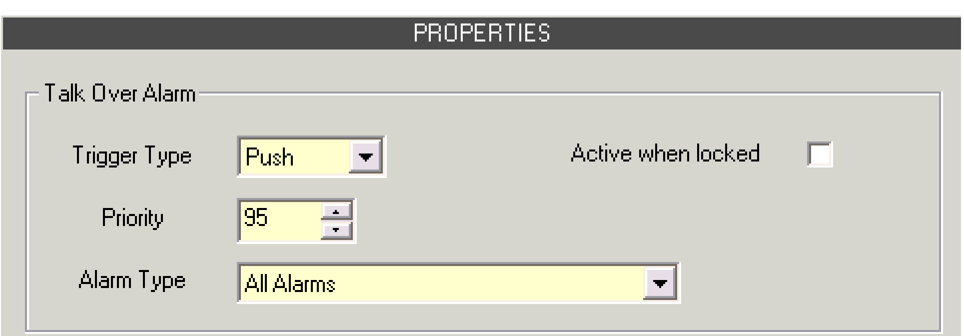
A switch of the type "Chime Key" allows the launch of a gong (chime) signal in the system.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the chime priority (0 to 100).
Chime Type	Select the desired type of gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Talk over Alarm Key

A switch of the type "Talk over Alarm Key" allows making an announcement during an alarm. During the announcement the alarm signal is off, and is started again after the announcement.



Element	Description
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states)
Priority	Select the priority (0 to 100) of the announcement. Must be higher than the alarm signal priority.
Alarm Type	Select the alarm type.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

On Key

A switch of the type “On” allows switching the PROMATRIX 8000 system on or off (standby) using a button of the call station.

Element	Description
Switch on priority	Select the priority (0 to 100) of the button.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Lock Key

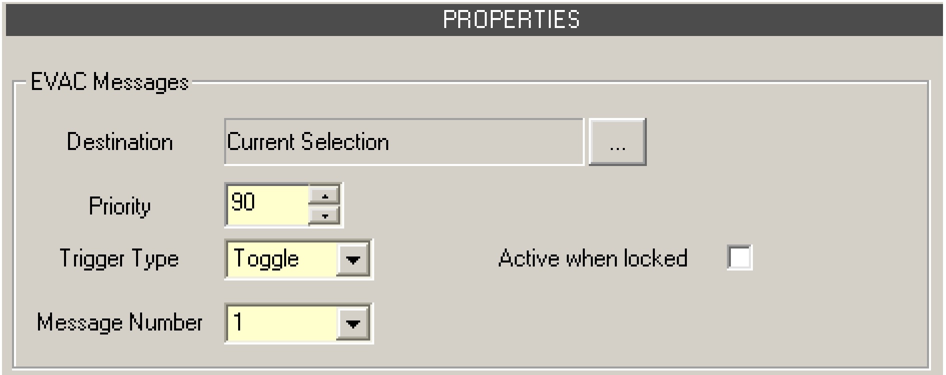
A switch of the type “Lock” allows locking the buttons of the call station. This button type can be assigned to a key switch only.

Notice!

If a button should stay active even if the call station is locked, the „Active when locked“ checkbox of this button has to be selected.

EVAC Message Key or Business Message Key

A switch of the type "EVAC Message Key" or “Business Message Key” allows starting a prerecorded message of type EVAC ore Business Message from the Message Manager.



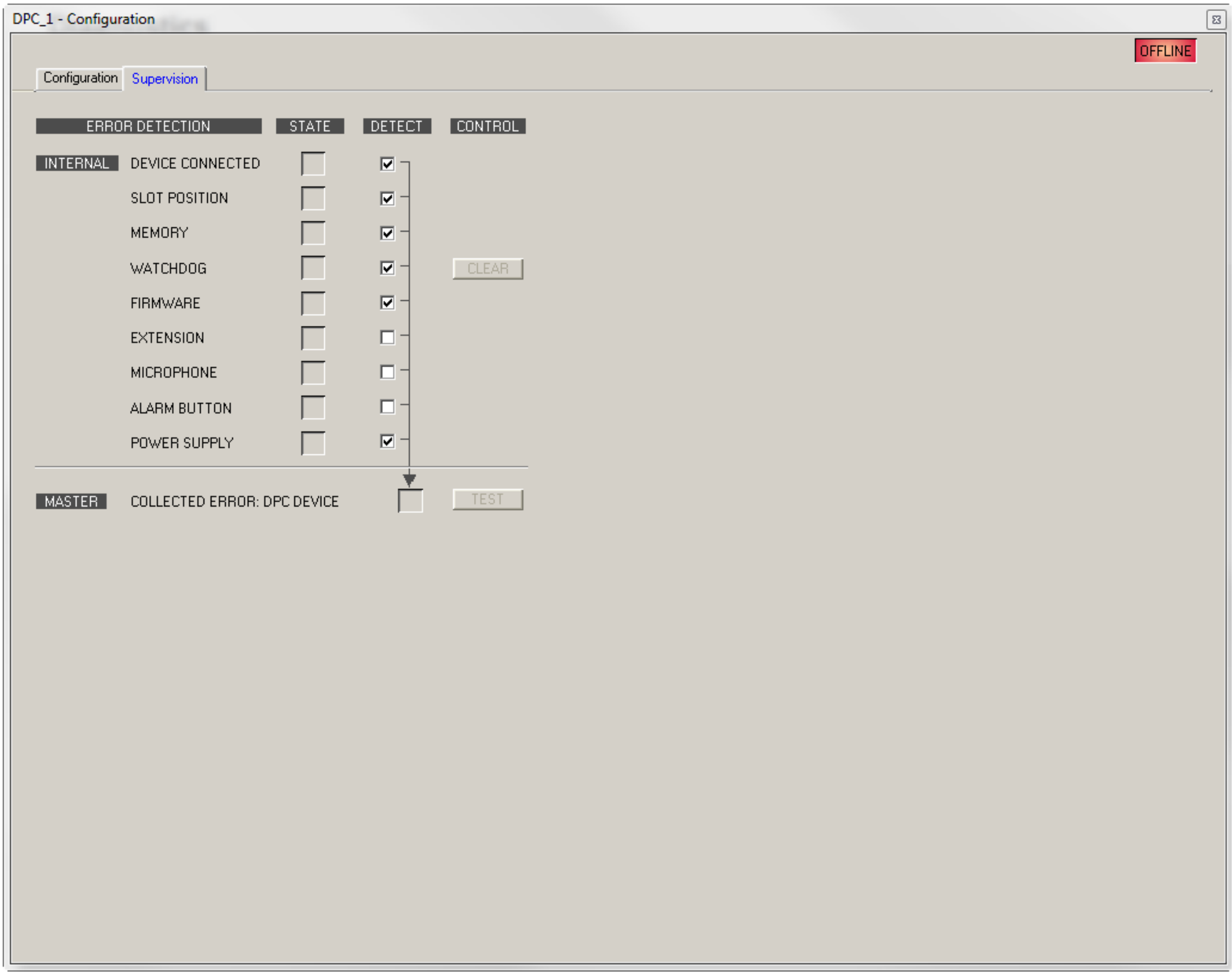
Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the priority (0 to 100) of the message.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger
Message Name	Select the message by name.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Loop	Select this checkbox to automatically repeat the selected message.

System Fault Ack/Res

A switch of the type “System Fault Ack/Res” allows to acknowledge or reset a system fault that is indicated at the call station. This type can be assigned to the DEL button only.

Diagnostics

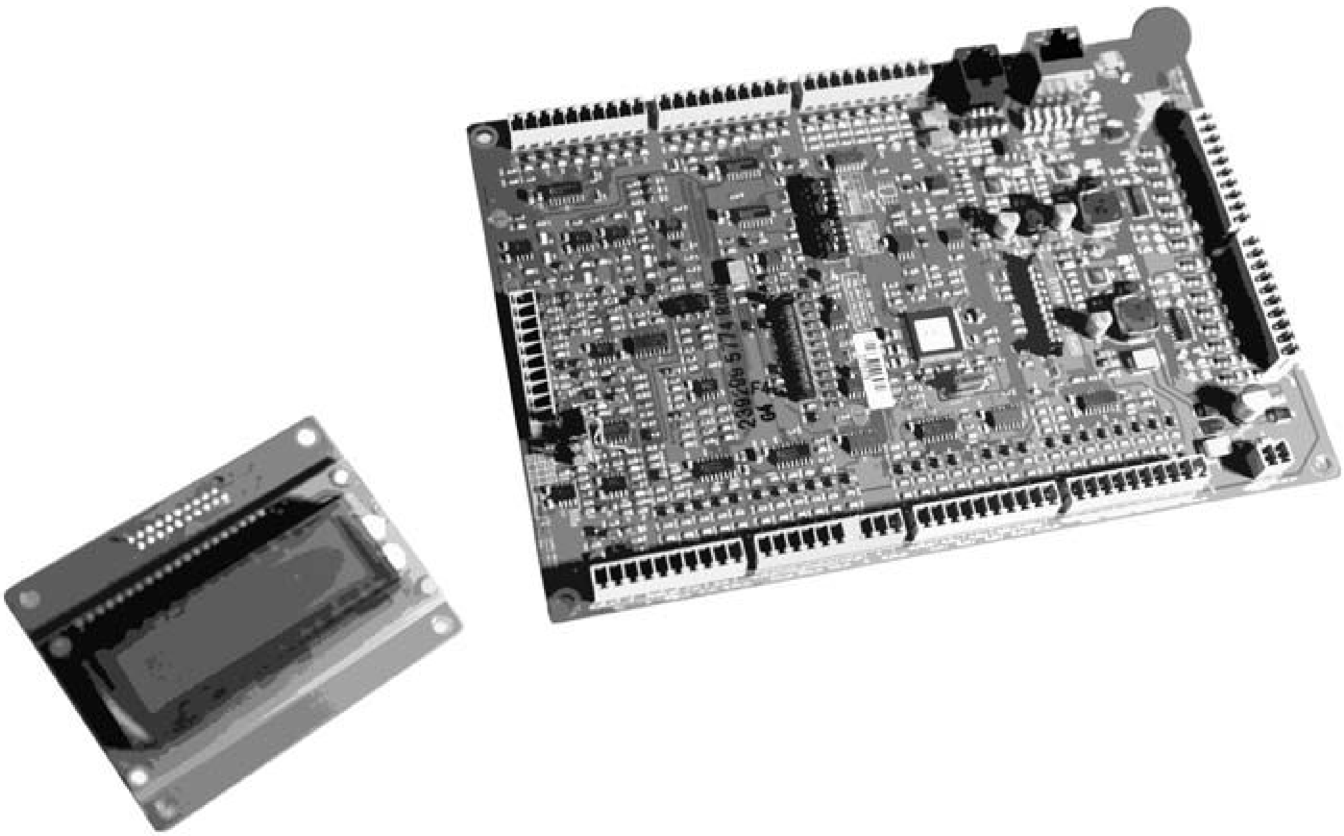
The diagnostics dialog is used for supervision of the call station.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time and the FAULT-LED at the DPC lights.
DEVICE CONNECTED	The PCA bus connection between DPM and DPC is broken.
SLOT POSITION	The call station is not connected to the correct slot position.
MEMORY	Memory error in DPC.
WATCHDOG + CLEAR	Watchdog error in DPC. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the DPC is to old.

EXTENSION	The number of call station extensions is to high or the addresses of the extensions are not correct.
MICROPHONE	Microphone error in DPC.
ALARM BUTTON	Supervision fault of the alarm button or the key switch.
POWER SUPPLY	Power supply out of range.
COLLECTED ERROR STATE: DPC DEVICE	The FAULT-LED at the call station lights at the occurrence of this type of error.

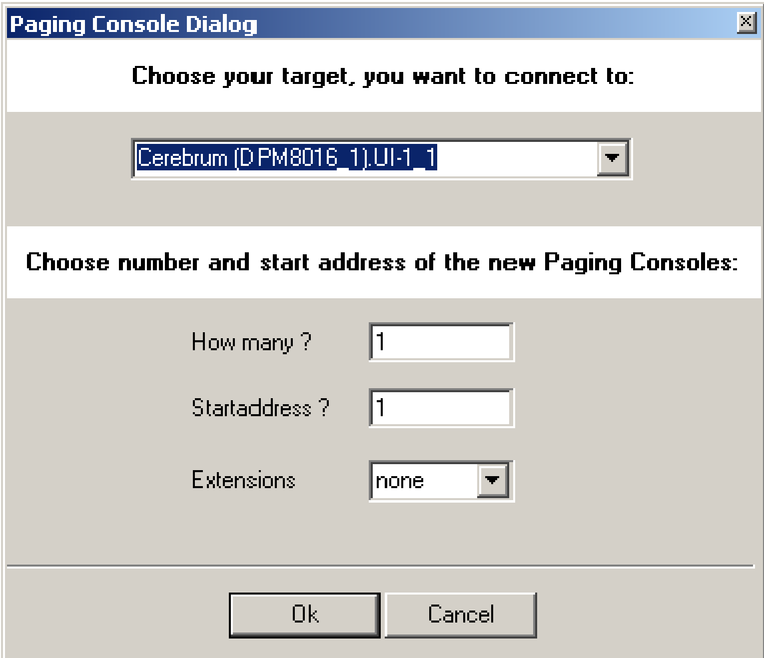
PMX-CSK Call Station Kit



The PMX-CSK is a call station kit for the PROMATRIX 8000 system. The call station kit allows connection of a permanently supervised microphone, a total of 20 buttons, a lighted LC-display and a loudspeaker. The 15 function keys can be permanently assigned for discrete zone addressing.

Additional PMX-CSK Features:
- Five pre-programmed function keys - LED for each button
- 15 freely programmable function/zone keys - Two LEDs for each button
- Internal supervision with event log - fulfills all requirements of relevant national and international standards

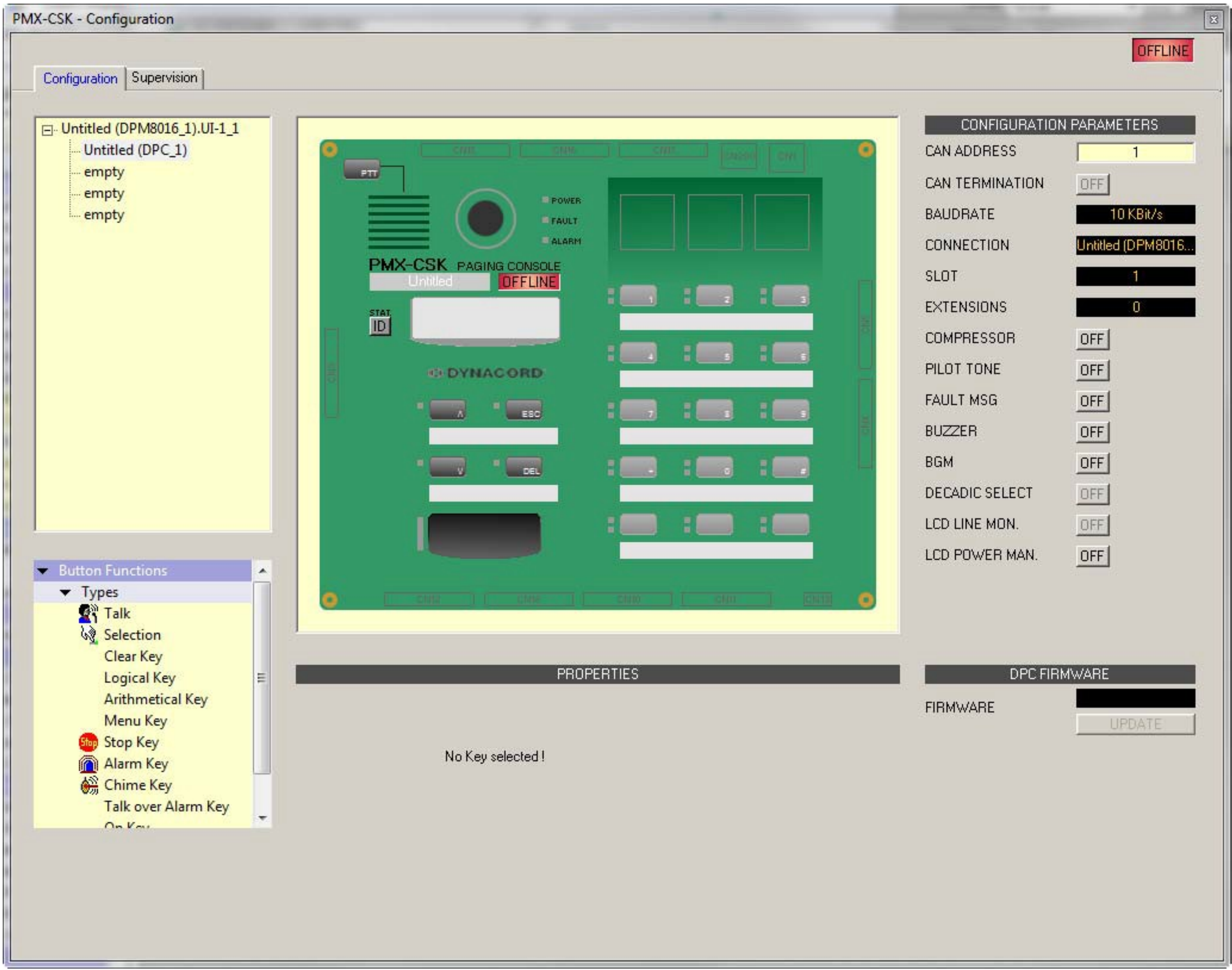
PMX-CSK Device
Start by creating an PMX-CSK Device in your IRIS-Net project. Drag an PMX-CSK from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

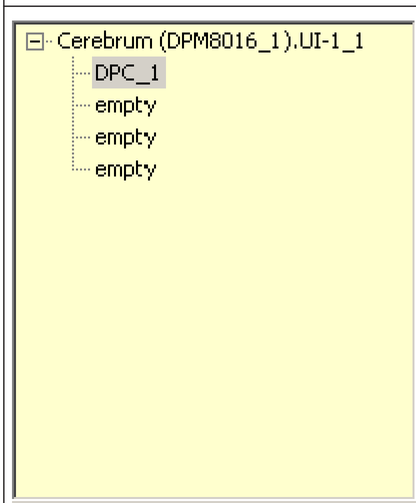


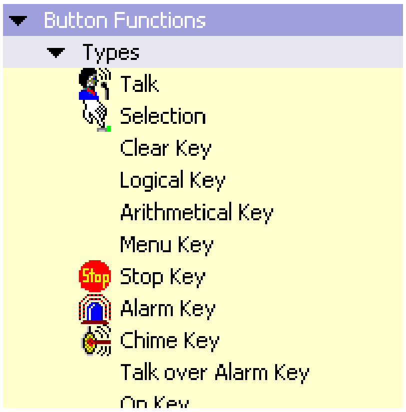
Select the UI-1 Universal Input Module of the DPM 8016 the call station is connected to. Specify the desired number of devices and the address of the call station. Click on the OK button to accept these settings.
The specified number of PMX-CSK devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an PMX-CSK device icon opens the configuration dialog window. Double clicking on a device for the first time will open the Configuration dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the PMX-CSK device icon.
The following table lists all available PMX-CSK dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	**Description**
Configuration	This window allows hardware settings to be configured, e.g. button configuration, network settings, device name.
Supervision	This window provides an overview of the operational state and current fault status of the call station.

Configuration
This page allows making basic settings and retrieve information, for example of button functions, network settings, device name, firmware version, etc.



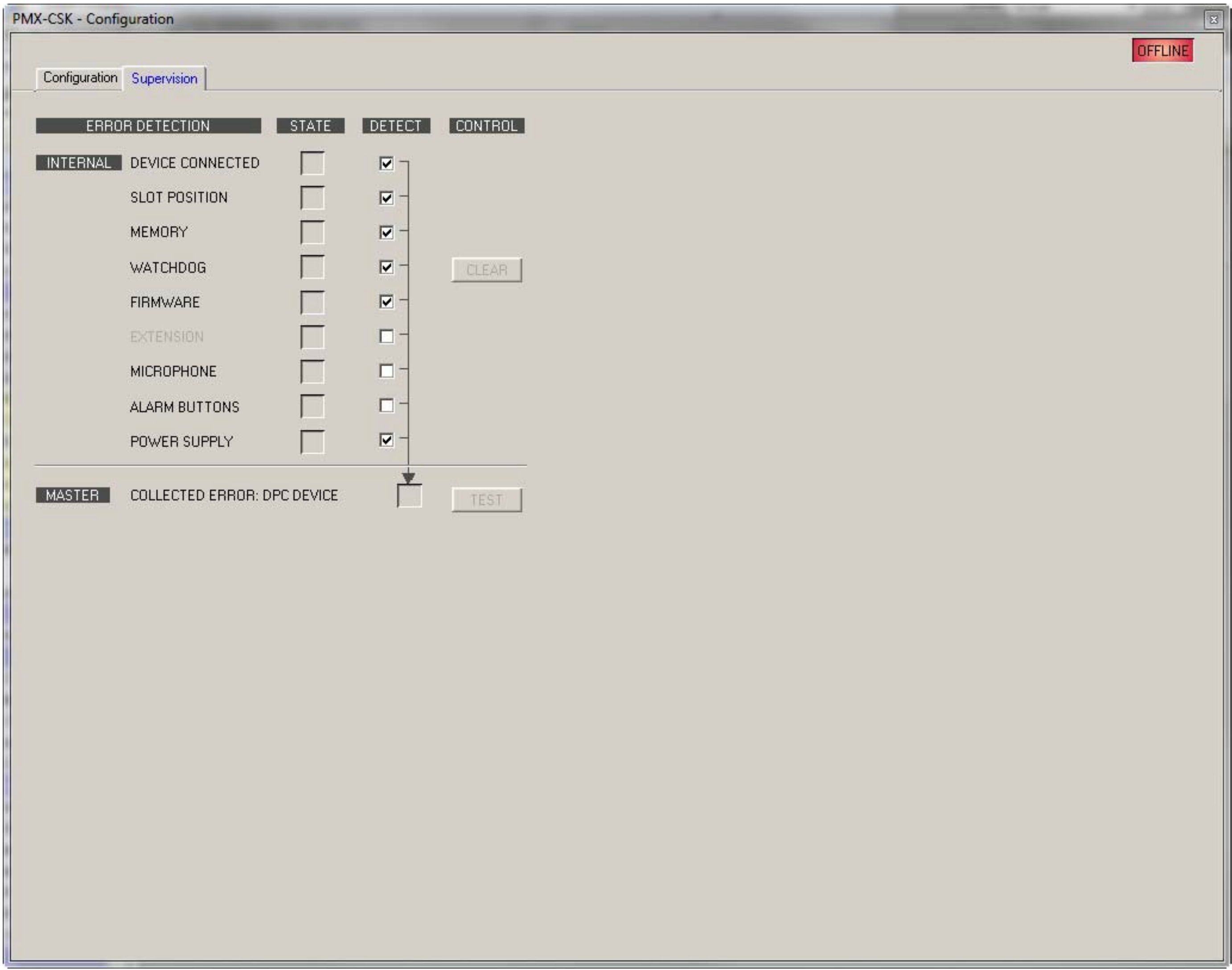
Element	Description
	This paragraph provides explanation of how to select the proper call station for configuration, when several call stations are connected to a single UI-1 Universal Input Module.

	Select the desired button type and drag it from this dialog box onto the button of the call station kit. Detailed information about different types of buttons is provided on the following pages.
ONLINE OFFLINE	The Online / Offline indicator signals whether the call station kit is included in the network or off-line. The red OFFLINE indicator signals that the corresponding call station is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding call station kit is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
CAN ADDRESS	Displays and lets the user enter the CAN address of the call station. Left-click in the ENTER field and enter the desired address in the range from 1 to 16. The entered value is adopted by pressing RETURN. The entered address has to match the setting of DIP switch S22 on the call station kit and may only exist once on the CAN- Bus. When adding new call stations to an IRIS-Net project, CAN addresses are automatically assigned in ascending order.
CAN TERMINATION	Please refer to the PMX-CSK owner‘s manual for details about CAN termination.
BAUDRATE	The baud rate of the call station kit. Defining the baud rate is performed via the UI-1 Universal Input Module of the DPM 8016.
CONNECTION	Name of the UI-1 module and the DPM 8016, the call station kit is connected to.
SLOT	Number of the slot, the UI-1 module is assembled to.
EXTENSION	Number of paging console extensions.
COMPRESSOR	Press this button (ON) to activate the internal compressor of the call station kit.
PILOT TONE	Press this button (ON) to activate the pilot tone supervision of the call station kit. **HINT: When using the pilot tone supervision only one call station can be connected to a PCA bus.**
FAULT MSG	Press this button (ON) if error messages should be indicated in the LC-display of the call station kit.
BUZZER	Press this button (ON) if errors should be signaled via the integrated buzzer.

BGM	Press this button (ON) if the BGM menu should be accessible in the LC-display of the call station kit.
DECADIC SELECT	Press this button (ON) to activate decadic zone selection via function/zone keys.
LCD LINE MON.	Press this button (ON) to activate the indication of line monitoring messages in the LC-display.
LCD POWER MAN.	Press this button (ON) to activate the indication of power management messages in the LC-display.
	When pressing this button, the backlight of the call station's LCD screen blinks regularly in quick succession. The status indicator of the call station Device in IRIS-Net blinks at the same time. This function serves for checking communication and for identification or search of a call station in a larger system.
FIRMWARE	Indicates the firmware version of the DPC when on-line.
UPDATE	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**

Supervision

The diagnostics dialog is used for supervision of the call station kit.



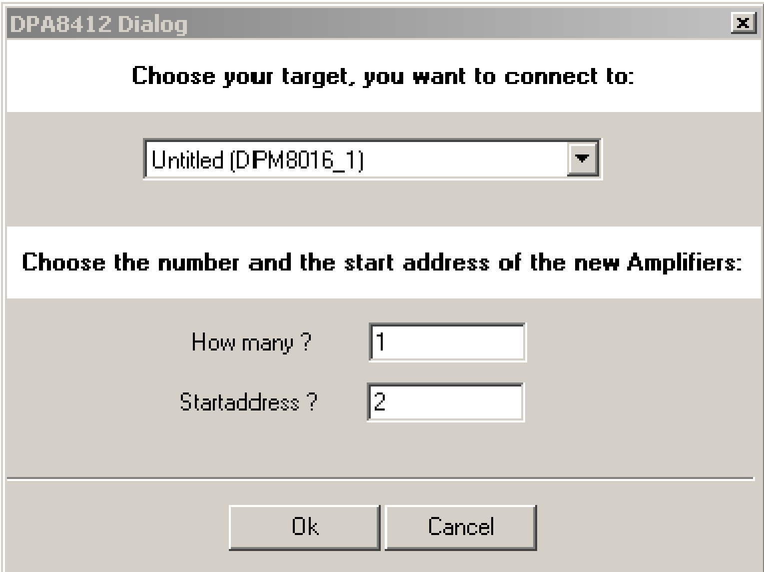
Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR: DPC DEVICE flag is set at the same time and the FAULT-LED output of the CSK is activated.
DEVICE CONNECTED	The PCA bus connection between DPM and CSK is broken.
SLOT POSITION	The call station kit is not connected to the correct slot position.
MEMORY	Memory error in CSK.
WATCHDOG + CLEAR	Watchdog error in CSK. This error type is logged according standards, press the button connected to ESC_K to reset the error.
FIRMWARE	The firmware version of the CSK is to old.
MICROPHONE	Microphone error in CSK.

ALARM BUTTONS	Supervision fault of the alarm button or the key switch.
POWER SUPPLY	Power supply out of range.
COLLECTED ERROR: DPC DEVICE	The FAULT-LED at the call station lights at the occurrence of this type of error.

DPA 8000 Power Amplifier

The DPA 8000 Series power amplifiers are Class-D amplifiers which have been designed for AC mains as well as for DC operation. The output voltage is galvanically isolated and constantly monitored for ground fault. A power saving mode and temperature controlled fans allow for energy efficient and nearly noiseless operation. Control and monitoring are carried out via the REMOTE CAN BUS. The amplifiers have been designed for use in an evacuation system. Control usually is taken care of by the DPM 8016 Paging Manager and the configuration by means of IRIS-Net. If amplifiers are operated without employing a DPM 8016, the configuration can be created in IRIS-Net and permanently be stored in the amplifier, so that the desired state (preset) is activated, each time the device is powered ON.
All power amplifiers have the following common features:
– Floating 100 V power outputs (internally configurable to 70 or 50 V)
– Amplifier blocks in Class-D technology
– Outputs are protected against idling and short-circuit
– Mains operation 230 V/120 V AC and/or 24 V DC emergency power
– Electronically balanced inputs
– Temperature monitoring
– Pilot tone and ground fault monitoring
– Fault message via floating READY contact
– All functions are processor-controlled
– Watchdog circuit for monitoring the processor system
– Nonvolatile FLASH memory for storing the configuration data
– Internal monitoring
– Integrated audio relays
– Line monitoring

DPA 8000 Device
Start by creating an DPA Device in your IRIS-Net project. Drag an DPA from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

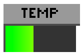
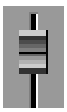
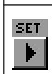
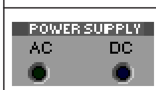


Select the DPM 8016 the amplifier is connected to. Specify the desired number of devices and the address of the amplifier. Click on the OK button to accept these settings.
The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

Control Panel

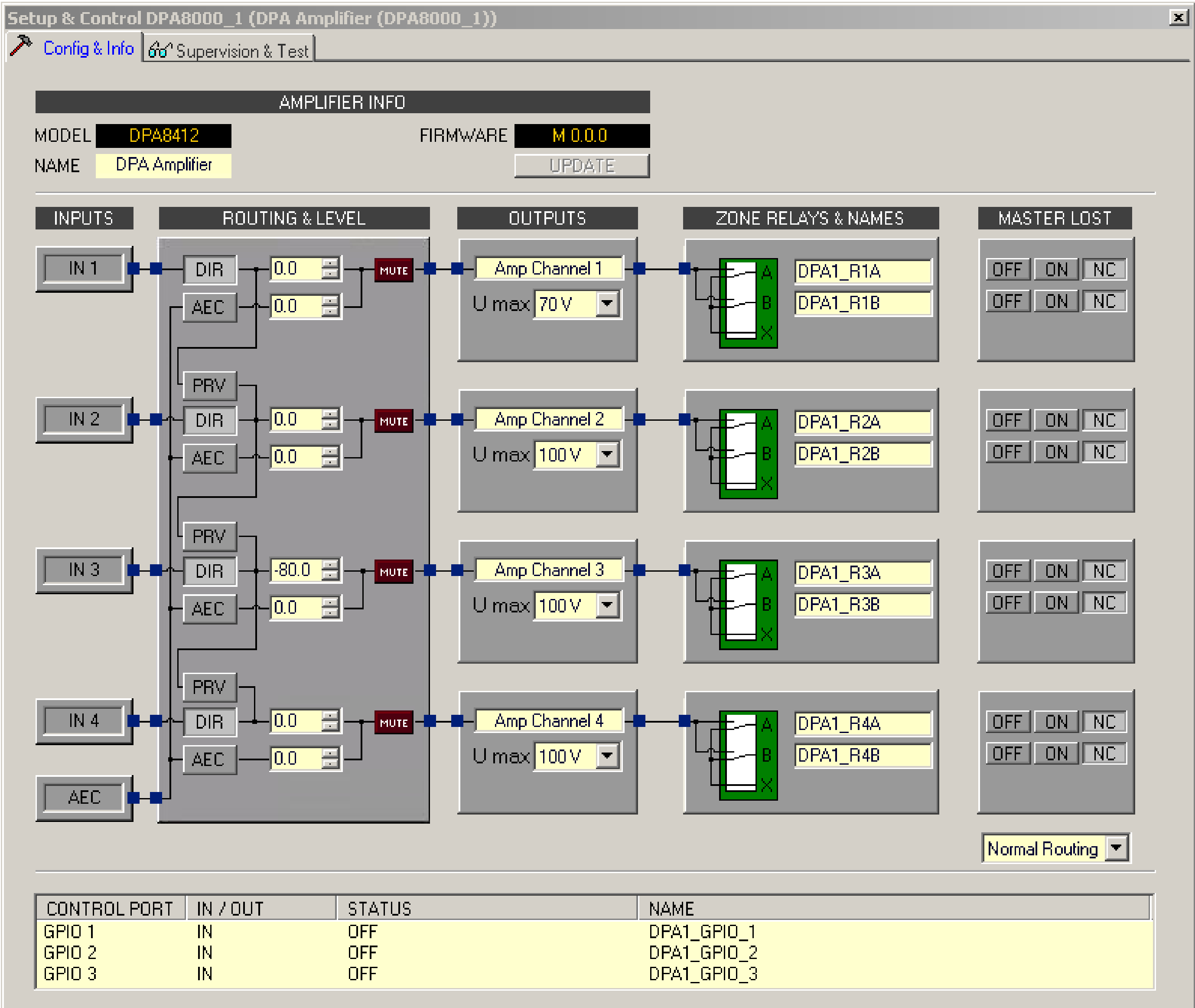
Double clicking with the left mouse button on an amplifier gets you to the Amplifier Control Panel, which provides access to the most important controls and indications of the selected amplifier. Simultaneously opening several Amplifier Control Panels and placing them in any order on the computer screen is possible as well. For dragging the panel windows around, please use the left mouse button and click on the title bar at the top of the window. Keep the mouse button pressed while dragging the panel.

Element	Description
DPA 8412	Amplifier Type (generated during amplifier selection or read from the amp while being on-line)
☒	Using the left mouse button, click on the Close button to close the Amplifier Control Panel.
Stage Left	A name can be assigned to each amplifier to specify its use or position. Click on the gray-shaded entry field below the Amplifier Type field and enter the desired name. Press Return on the keyboard to acknowledge the entered name. **HINT: Entering amplifier names is also possible within the Setup & Control Panel on the Config & Info page. CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
ONLINE OFFLINE	The Online / Offline indicator signals whether the selected amplifier is included in the network or off-line. The red OFFLINE indicator signals that the corresponding amplifier is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding amplifier is on-line and that sending and receiving data is possible.
1 Amp Channel 1	The amplifier channels are named channel 1 to 4, depending on the amplifier type. A name can be assigned to each channel to easily identify its allocation and use. Using the left mouse button, click in the entry field and enter the desired name for the channel. Press Return on the key- board to acknowledge your entry.

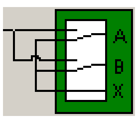
	The TEMP display shows the amplifier's internal temperature as a graph. The indicator lights green whenever the amplifier is operated in its nor- mal operational temperature range. The indicator lights yellow whenever the amplifier builds up heat because of continuous high output. However, since the internal fans provide sufficient ventilation there is no risk of thermal overload in this state. As soon as temperature indication changes to red, reducing the output level is strongly recommended. Otherwise the amplifier might cease operation because of thermal overload.
 	The level controls are for adjusting the overall amplification of the DIRECT IN input signal of the corresponding amplifier channel. Setting the level controls to a value of 6dB provides full output capacity. The numerical field below the level controls indicates the set level, by which the output amplification is attenuated, in dB. **HINT: This level control does not indicate the level of the AEC input, even if the AEC input routing is selected in the Config&Info window. Use the numerical field available in the Config&Info window to set the AEC input level.**
 	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
	Clicking this switch activates the STATUS indicator on the amplifier's rear panel as well as in the amplifier's front panel window in the IRIS-Net software. Normally, the STATUS indicator blinks only during serial communication. Once the STATUS switch is engaged, the STATUS indicator blinks in a steady but fast sequence. This function is meant for checking communication and for identifying or searching an amplifier in a large system setup.
	The address field indicates the set amplifier address. Assigning a new address is also possible by clicking into the field with the left mouse but- ton and entering the desired amplifier address. Available values are 1 to 250. Press Return on the computer keyboard to acknowledge your entry. The assigned address and the address specified by the setting of the selection switch on the amplifier's rear panel have to be identical. Each address can exist only once within a system.
	Clicking on the SET button opens the Setup & Control Window, which provides access to all amplifier- and DSP-parameters, control and monitoring functions plus additional function groups.
	The AC LED lights if mains power is available. The DC LED lights if battery power is available.
	This soft-key allows switching an amplifier on or off. The STANDBY and POWER indicators signal the actual operational status.
	These indicators show the amp's actual operational status. STANDBY lights whenever the amplifier is in stand-by mode. POWER lights whenever the amplifier is powered-on and ready for operation. If neither one of the indicators lights, the amplifier is either off-line or powered-off.

Config & Info

The Config & Info window provides information and basic settings for the selected amplifier. To select the page click on the Configuration & Information tab in the Setup & Control Window.

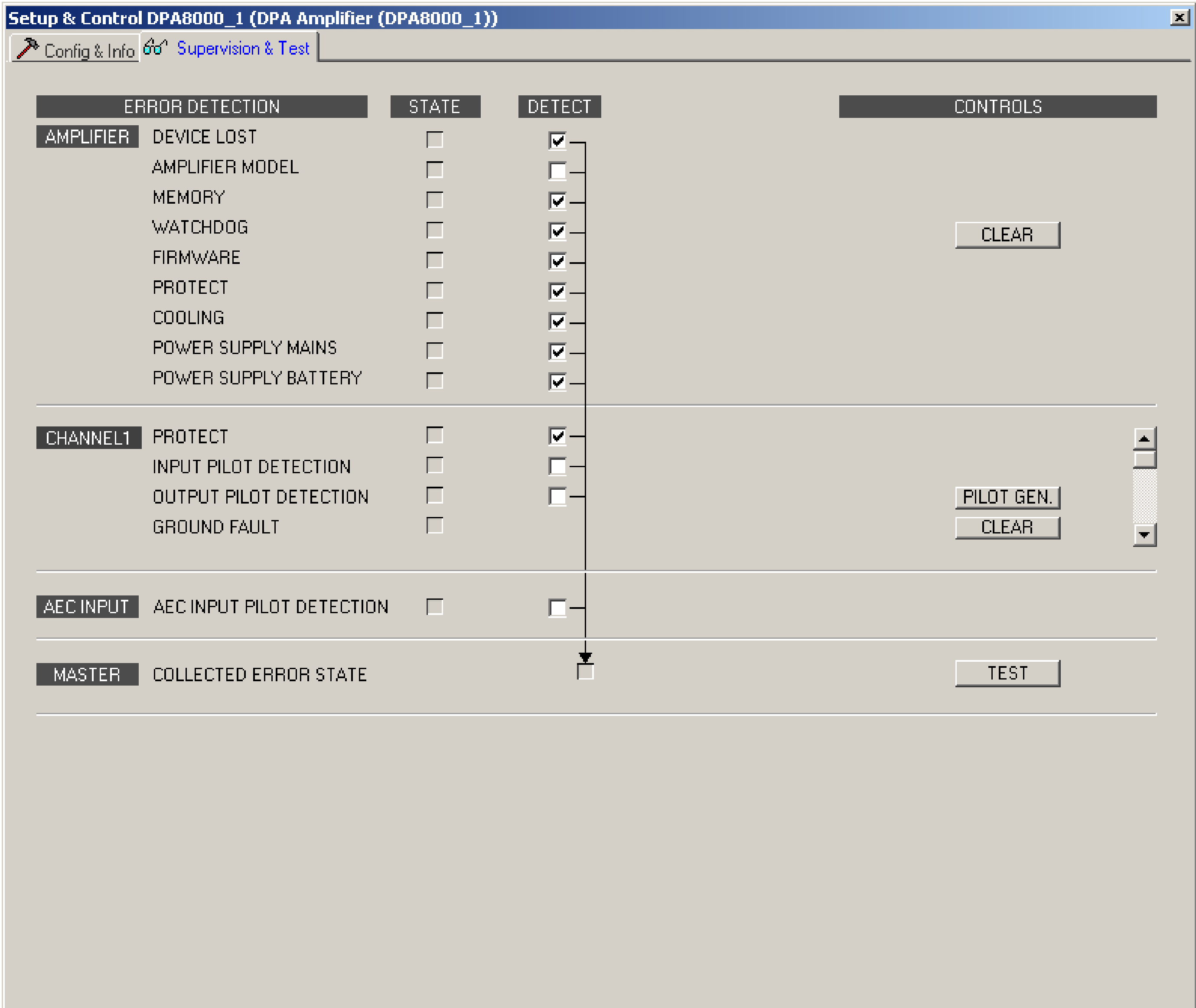


Element	Description
MODEL	Shows the amplifier type
NAME	IRIS-Net internal name of the amplifier
FIRMWARE	Shows the amplifier‘s software version number (operating system, firmware)
UPDATE	Opens the firmware update dialog.
PRV / DIR / AEC	For each amplifier output channel the signal of the direct input (DIR), the AEC input (AEC) or the previous input (PREV) can be selected.
0.0	The numerical field of the DIRECT input is identical to the numerical field below the level controls in the Amplifier Control Panel. So the field indicates the actually set attenuation, by which the internally specified amplification is attenuated, in dB. The numerical field of the AEC input allows to set the attenuation of this input independently.

MUTE	The MUTE button is for attenuating the output level of the corresponding amplifier output to -∞. Clicking the MUTE button with the left mouse button mutes the corresponding amplifier output. The MUTE button is virtually pressed and lights red. Clicking the MUTE button once again with the left mouse button disables the mute-function and the amplifier output is again active. The MUTE button is virtually disengaged and not lit.
Amp Channel 1	The text field allows specifying a name for the corresponding output channel.
100 V	The output voltage of the output channel can be set to 50, 70 or 100 Volts. The configuration in IRIS-Net must be identical to the hardware configuration of the amplifier, please refer to the owner‘s manual for details.
	For each output channel, the output relays A and B can be switched between the input signal of the channel and the signal of the input X.
DPA1_R1A	The text field allows specifying a name for the corresponding output relay.
Normal Routing	Select the input signal to be used if the connection to the DPM is lost: – Normal Routing: The current setting (PREL. / DIRECT /AEC) will not be changed if the connection is lost. – AEC Routing: For all output channels the signal of the AEC input will be used.
OFF / ON / NC	Select the preferred status of the output relay if the connection to the DPM is lost: OFF: Relays open ON: Relays closed NC: Status of relays does not change
CONTROL PORT	This provides a listing of the three control ports.
IN / OUT	Select if the port should be used as control input (IN) or as control output (OUT).
STATUS	In online mode the state of the port is indicated. When used as control input: – ON: Input voltage below 5 Volt. – OFF: Input voltage above 10 Volt. When used as control output: – ON: The output is closed to ground. – OFF: The output has high impedance.
NAME	The text field allows specifying a name for the control port.

Supervision & Test

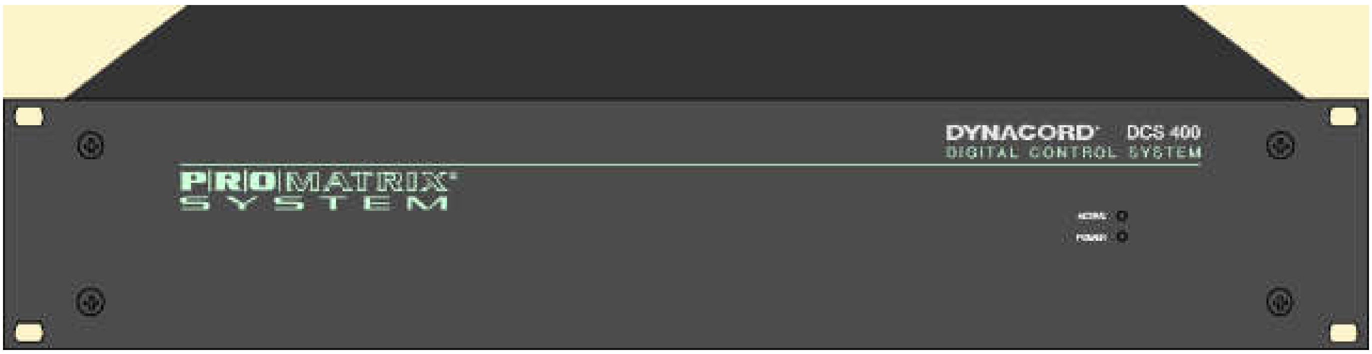
The Supervision & Test Dialog integrates functions for testing and monitoring power amps. You have the option to choose, which errors are combined and indicated in a general fault message.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time and the FAULT-LED at the DPA front panel lights. Additionally the ready relays changes to „not operational“ state.
AMPLIFIER	
DEVICE LOST	The CAN bus connection between DPM and DPA is broken.
AMPLIFIER MODEL	Amplifier type in IRIS-Net and actually connected hardware do not match.
MEMORY/DATA	Memory or Read/Write error.

WATCHDOG	Watchdog error in DPA. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the DPA is not compatible with the DPM firmware version. A firmware update is recommended.
PROTECT	When the red PROTECT / REDUCTION indicator lights, one of the internal protections has been activated.
COOLING	Temperature overload of the DPA.
POWER SUPPLY MAINS	Power supply mains error.
POWER SUPPLY BATTERY	Power supply battery error.
CHANNEL1	
PROTECT	When the red PROTECT / REDUCTION indicator lights, one of the internal protections of the output channel has been activated.
INPUT PILOT DETECTION	This indicator lights when the pilot tone at the DIRECT input is missing.
OUTPUT PILOT DETECTION	This indicator lights when the pilot tone at the amplifier output is missing.
PILOT GEN.	The PILOT GEN button allows activating the pilot tone generator of the amplifier channels. Default: Deactivated, 6 V, 19600 Hz The property: PilotHighGain=1 allows to increase the pilot tone level to 12 V. The property: PilotTone.Freq allows adjusting the frequency of the generated signal in the range 15000 Hz to 25000 Hz.
GROUND FAULT	A ground fault error at the amplifier output occurred. Press the CLEAR button to reset the error.
AEC INPUT	
AEC INPUT PILOT DETECTION	This indicator lights when the pilot tone at the AEC input is missing.
MASTER	
COLLECTED ERROR STATE	The FAULT LED at the DPA front panel lights at the occurrence of this type of error.
TEST	Press this button to activate the COLLECTED ERROR STATE manually.

DCS Digital Control System



Introduction

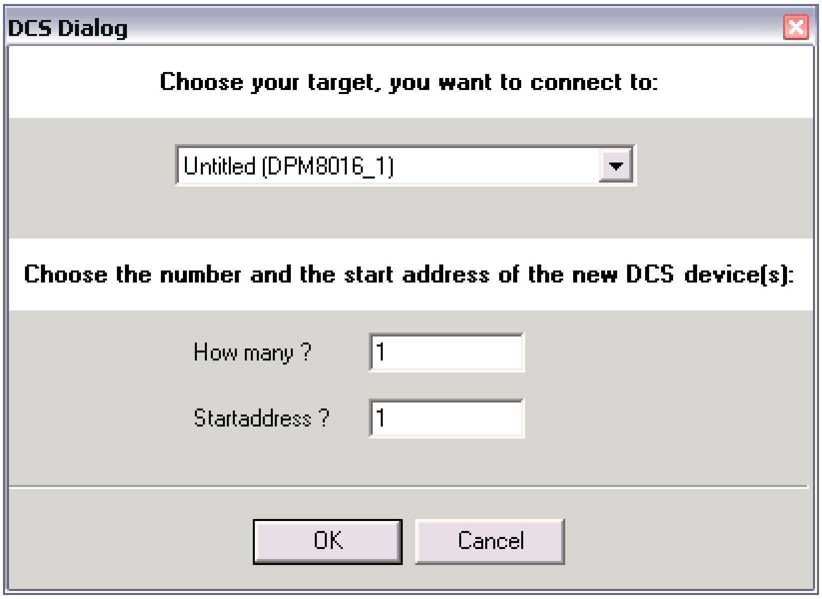
DCS 400 Series modules expand the system control possibilities in an IRIS-Net system in many respects. Expanding the DCS system is also possible after installation, in case additional functions are needed. A DCS system consists of a single, two or three DCS 400 frames which can host different relay boards, logic input boards, and analog level input/output boards. Connecting the DCS system with an IRIS-Net system controller is always accomplished using a DCS 801R controller module which gets connected via CAN-Bus to an IRIS-Net system controller. The maximum number of DCS 801R modules that can be connected to an IRIS-Net system controller is 15; i.e. an IRIS-Net system controller can maximally host 15 individual DCS systems. The complete configuration of a DCS system is created in IRIS-Net, where you define the number and type of DCS cards to be used in the system as well as the functions of individual inputs and outputs. The procedure of creating a configuration as well as the different options available during this process is explained in detail in the following chapters.

DCS Device

First of all, create a new DCS device in your IRIS-Net project by dragging a DCS with the desired number of height units (HU) from the Object List's Devices category or out of the Devices Window into the worksheet (please also refer to the chapter "Configuration of Devices and Menus").

HINT: Before being able to add a DCS Device, a DPM 8016, N8000 or P 64 have to exist in the project already. Only 1 U units shall be used for EN 54 compliant systems.

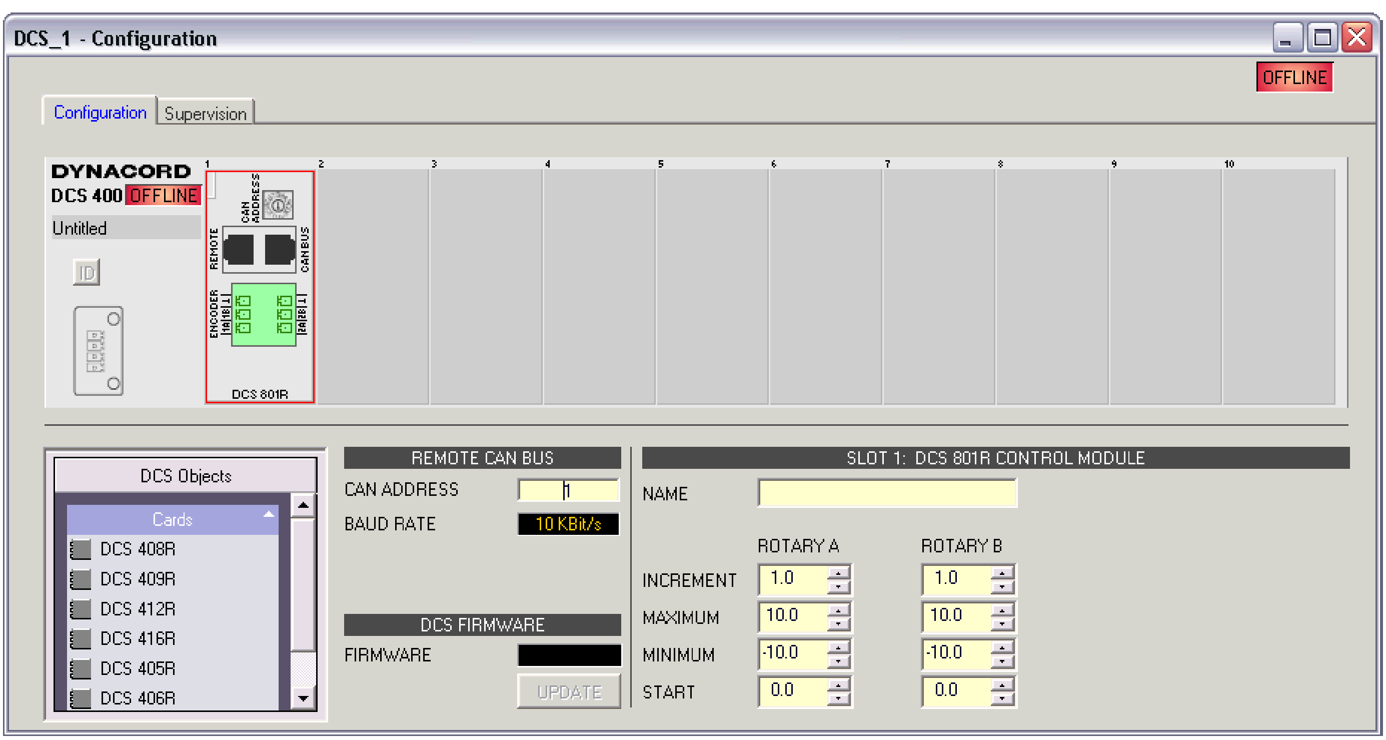
Always make sure to add a DCS with a sufficient number of card slots in your IRIS-Net project. Subsequent "heightening" of a DCS, e.g. from DCS 1HU to DCS 2HU, is not possible. The following dialog appears:



Please specify the number of desired devices (max. 15), the CAN address of the first DCS system (1 to 15), and the communication interface. Confirm your settings with OK. The desired number of DCS Devices appears in the worksheet. The devices can be marked and freely arranged and positioned within the worksheet.
Double clicking on a DCS Device opens the configuration dialog box.

Configuration dialog

Double clicking on a DCS Device in the IRIS-Net worksheet opens the Configuration Dialog box which allows making all the settings that are necessary for starting up the DCS system.



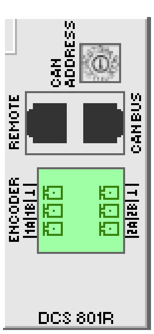
Element	Description
CAN ADDRESS 1	This field indicates and allows entering the DCS system address. Left-click in the input field and enter the desired address in a range from 1 to 15. Press "Return" to apply the address. The entered address has to correspond to the setting of the address selection switch on the front panel of the DCS 801R. The DCS system address is exclusive; i.e. it may exist only once in the system. When creating new DCS Devices in IRIS-Net, addresses are automatically assigned in ascending order.
BAUD RATE n/a	Indicates the CAN bus transfer rate when in on-line mode.
FIRMWARE	Indicates the DCS firmware version when in on-line mode.
UPDATE	Opens the firmware update dialog.

EQUIPMENT
A DCS system may consist of a single, two or three DCS 400 frames. Drag & dop cards from the DCS Objects list to empty slots to equip the DCS 400 frame. Right click a slot to open the context menu. The context menu includes commands to equip, delete or move cards.
When equipping these frames, the following rules need to be observed:

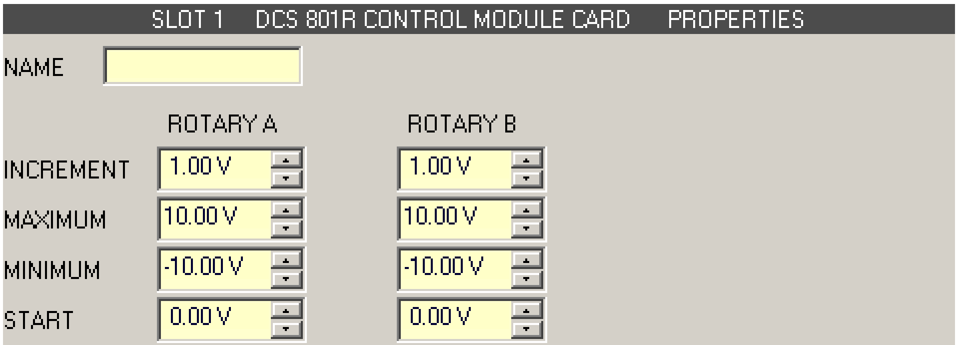
- Frame 1: Slot 1 always has to be equipped with a DCS 801R. IRIS-Net automatically adds a DCS 801R. Rearranging or deleting the DCS 801R is not possible. If the DCS system consists of only one frame, slots 2 to 10 can be equipped with the following card types: DCS 408R, DCS 409R, DCS 412R, DCS 416R, DCS 406R. If a second frame is included, the last slot from the left has to be equipped with a DCS 405R.
- Frame 2: Slot 1 always has to be equipped with a DCS 405 R. If the DCS system consists of only two frames, slots 2 to 10 can be equipped with the following card types: DCS 408R, DCS 409R, DCS 412R, DCS 416R, DCS 406R. If a third frame is included, the second frame has to be equipped with a DCS 405R (like in the first frame).
- Frame 3: Slot 1 always has to be equipped with a DCS 405 R. Slots 2 to 10 can be equipped with the following card types: DCS 408R, DCS 409R, DCS 412R, DCS 416R, DCS 406R.
- Number and type of cards in a DCS system: A DCS system can maximally be equipped with 17 relay modules (DCS 408R / DCS 409R), 5 logic input modules (DCS 412R) or 5 analog input/output modules (DCS 416R) respectively.
- Gaps between individual cards are not allowable. If necessary, gaps have to be filled using cards of the type DCS 406R.

DCS card types

DCS 801R CONTROLLER MODULE

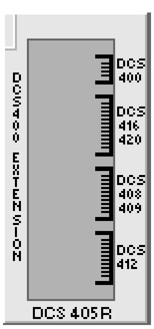


The DCS 801R module is used as an interface for relay boards, logic input boards, analog level input/output boards, and rotary encoders. Control takes place via REMOTE CAN BUS interface. A DCS 801R control module can maximally host 17 relay modules (DCS 408R /DCS 409R), 5 logic input modules (DCS 412R), 5 analog input/output modules (DCS 416R), and 2 rotary encoders.



Element	Description
NAME	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible**.
INCREMENT 1.00 V	Rotary encoder step size (voltage change)
MAXIMUM 10.00 V	Maximum voltage of rotary encoder
MINIMUM -10.00 V	Minimum voltage of rotary encoder
START 0.00 V	Rotary encoder voltage after power-on

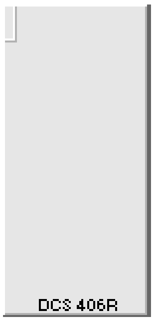
DCS 405R EXTENSION MODULE



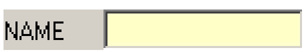
The DCS 405R module is used for connecting one DCS-400 frame to another DCS-400 slide-in card and/or for adapting additional DCS-400 cards. This can be advantageous in situations, where the number of modules that a DCS 801R is equipped with exceeds the maximum number of modules that a DCS-400 rack can host.

Element	Description
NAME	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

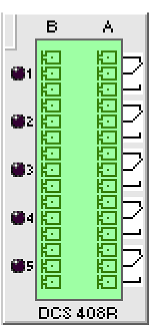
DCS 406R SHIELDING MODULE



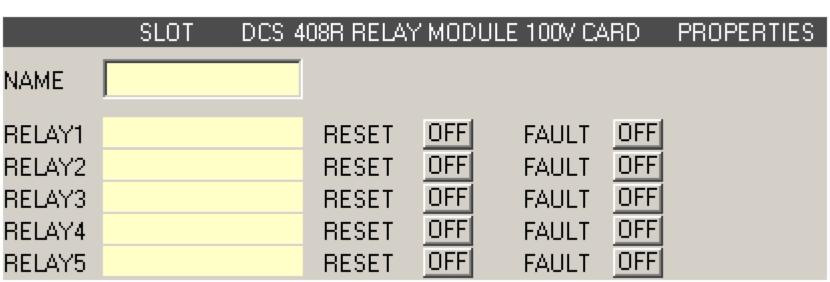
The DCS 406R module is used as a shielding between 100 volts modules (DCS 408R) and low-voltage modules.

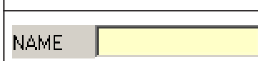
Element	Description
	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

DCS 408R LINE / AF RELAY MODULE



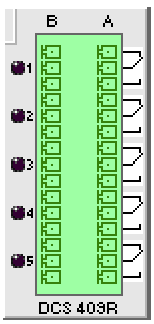
The DCS 408R module offers relay-switching contacts for the switching of audio signals or other control functions. The module is mainly used for switching 70 volts or 100 volts loudspeaker lines. It can also be used to provide switching contacts when higher voltages and/or currents exist.



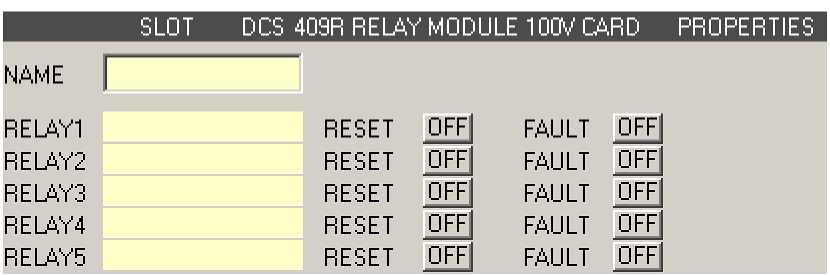
Element	Description
	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

RELAY1	Name of the relay **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
RESET OFF	State of the relay during a RESET of the DCS system. The DCS system is reset for example after changing the configuration or after a power outage.
FAULT OFF	State of the relay in error condition. Detailed information about the "error condition" of a DCS system is provided in the description of the DCS 801R module.

DCS 409R LINE / AF RELAY MODULE

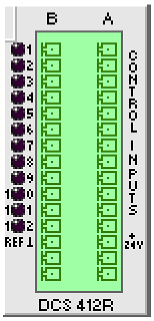


The DCS 409R module offers relay-switching contacts for the switching of (line level) audio signals or other control functions.

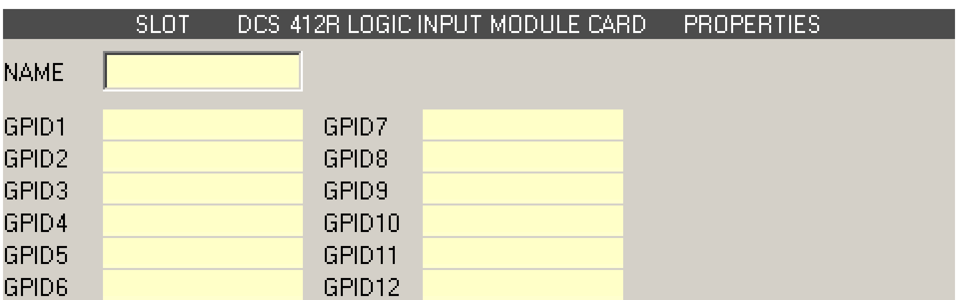


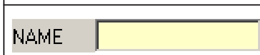
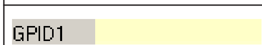
Element	Description
NAME	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
RELAY1	Name of the relay **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
RESET OFF	State of the relay during a RESET of the DCS system. The DCS system is reset for example after changing the configuration or after a power outage.
FAULT OFF	State of the relay in error condition. Detailed information about the "error condition" of a DCS system is provided in the description of the DCS 801R module.

DCS 412R LOGIK INPUT MODULE

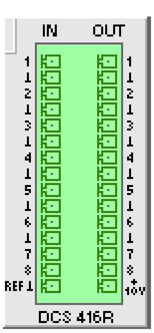


The DCS 412R module is used for connecting control lines, buttons, switches, and sensors to be able to evaluate their statuses (on, off) in the IRIS-Net system.

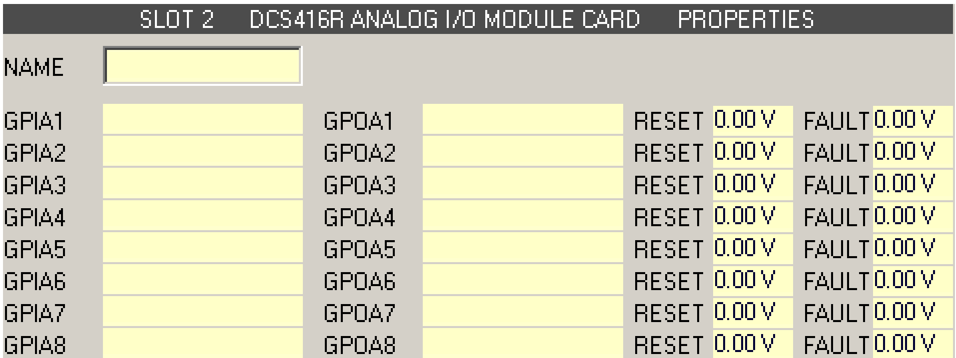


Element	Description
	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
	Name of the digital input **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**

DCS 416R ANALOG INPUT / OUTPUT MODULE



The DCS 416R module offers analog inputs and outputs for them to be used in control and monitoring functions. Voltages from 0 to 10 V DC with 256 different levels can be applied to the inputs as well as to the outputs. Potentiometers for volume control can be connected using the reference voltage output.

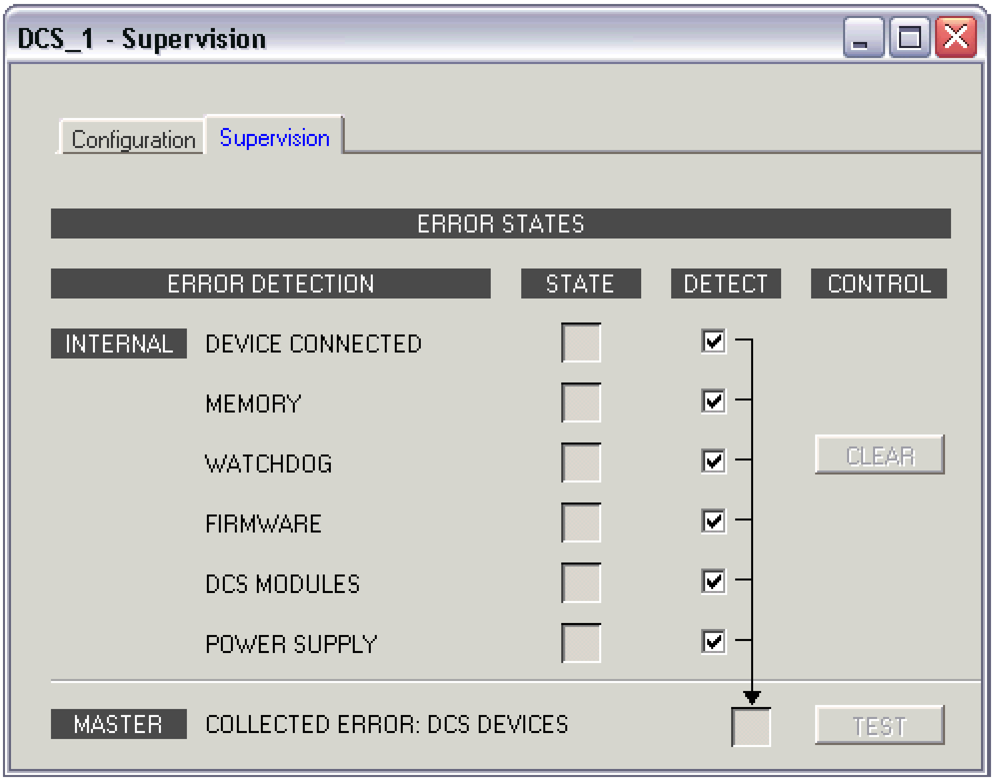


Element	Description
NAME	Name of the module **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
GPIA1	Name of the analog input **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
GPOA1	Name of the analog output **CAUTION: Using * (asterisk) and/or = (equal) signs in a name is not permissible.**
RESET 0.00 V	Voltage at the analog output during a RESET of the DCS system. The DCS system is reset for example after changing the configuration or after a power outage.
FAULT 0.00 V	Voltage at the analog output in error condition.

Supervision Dialog

The Supervision dialog is used for supervision of the DCS. You have the option to choose, which errors are combined and indicated in a general fault message.

HINT: This Supervision dialog is only available when the DCS is connected to a DPM 8016.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR: DCS DEVICES flag is set at the same time.
DEVICE CONNECTED	The CAN bus connection between DPM and DCS is broken.
MEMORY	Memory or Read/Write error.
WATCHDOG	Watchdog error in DCS. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the DCS is not compatible with the DPM firmware version. A firmware update is recommended.
DCS MODULES	The module configuration of the DCS in IRIS-Net does not match the actual module configuration.
POWER SUPPLY	Supply voltage out of range.
COLLECTED ERROR: DCS DEVICES	The state of this error is forwarded to the DPM 8016, see *Supervision Dialog, page 718*.
TEST	Press this button to activate the COLLECTED ERROR: DCS DEVICES manually.

PROMATRIX 6000

PMX-4CR12

The PMX-4CR12 controller is the central paging manager for the PROMATRIX 6000 system. Eight local audio inputs can be switched to four audio outputs. A two channel message manager is integrated. The controller provides all the audio processing, supervision and control functions for a complete PROMATRIX 6000 system. A single controller supports up to 16 call stations and 492 paging zones. The controller is equipped with 12 zones, 18 GPIs and 19 GPOs. One controller can handle up to 2000 W loudspeaker load. Additional zones and power can be added by using up to 20 external routers and 40 amplifiers with each 2 × 500 W. The zone indicator lights on the front indicate the current status of every zone:

- Green: Zone in use for non emergency purpose
- Red: Zone in use for emergency purpose
- Yellow: Zone fault detected
- Off: Zone in idle condition

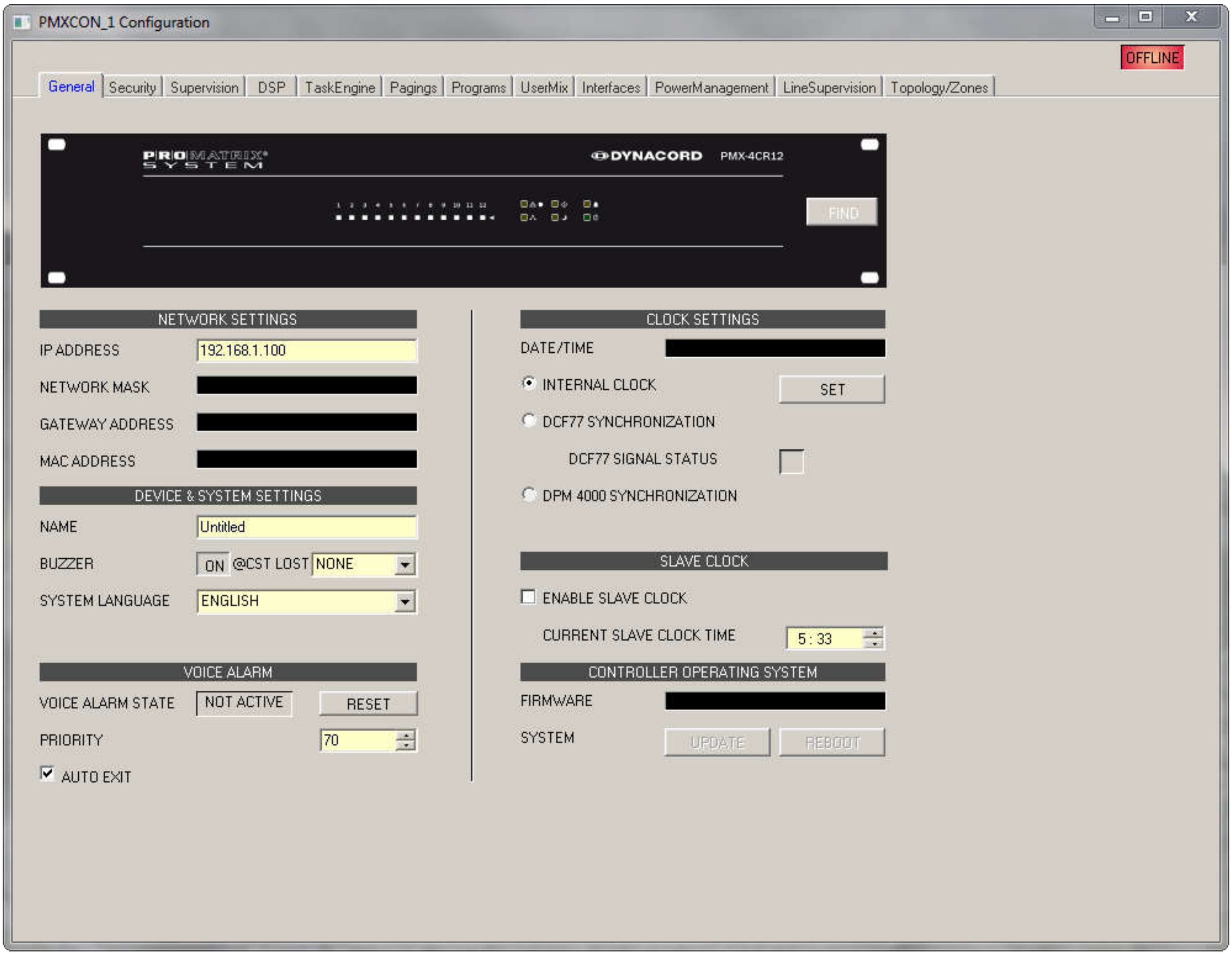
PMX-4CR12 Device

Start by creating a Controller Device in your IRIS-Net project. Drag a Controller from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu).
Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.
The following table lists all available device dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	**Description**
General	This window allows hardware settings to be configured, e.g. network settings, device name, system time and firmware version.
Security	This window allows editing passwords.
Supervision	This window provides an overview of the operational state and current fault status of the device.
DSP	This window allows editing the DSP configuration of the device.
Task Engine	This window lets you configure the Task Engine of the device.
Pagings	This window lets you configure dynamic add/sub zones (VAR pattern).
UserMix	This window lets you configure background music.
Interface	From this window the interfaces (e.g. CAN bus, GPIO control port) can all be configured. **HINT: Ethernet interface settings are edited under General dialog in the paragraph Network Settings.**
Power Management	From this window the power management of the device can be configured.
LineSupervision	The line supervision of the device can be controlled and supervised from this window.
Topology/Zones	This windows lets you configure topologies and zones of the system.

General Dialog

Double clicking on a PMX-4CR12 by default opens the General dialog box. Here, the user can make basic settings that are necessary for flawless operation. All elements of the displayed PMX-4CR12 front panel are active in on-line mode and correspond to the actual indicators on the unit.

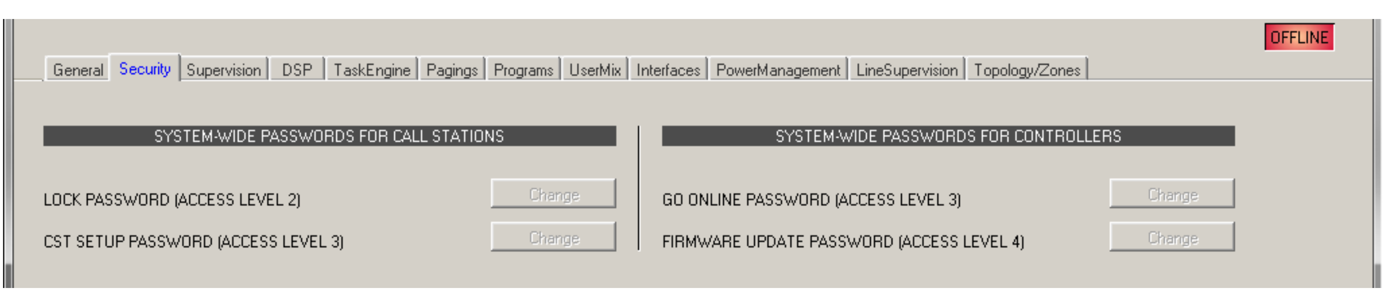


Element	Description
IP ADDRESS	Indicates the IP address of the PMX-4CR12’s Ethernet port (factory setting: 192.168.1.100). Enter the address of the PMX-4CR12 with which you want to establish on-line communication.
NETWORK MASK	Indicates the Ethernet port’s network mask (factory setting: 255.255.255.0).
GATEWAY ADDRESS	Indicates the standard gateway of the Ethernet port (factory setting: 192.168.1.1).
MAC ADDRESS	Indicates the MAC address of the connected PMX-4CR12 when on-line. The MAC address of the PMX-4CR12 is also shown on a label on the unit’s rear panel.

NAME	IRIS-Net internal device name of the PMX-4CR12.
BUZZER	Select ON to indicate a connection failure to a call station (selectable via the drop down field) via the integrated buzzer of the PMX-4CR12.
SYSTEM LANGUAGE	Select the system language of the PROMATRIX 6000 system.
VOICE ALARM STATE	This indicator shows “ACTIVE” if the device is in voice alarm state, else “NOT ACTIVE”.
RESET	Press the RESET button to deactivate the voice alarm state.
PRIORITY	Select the priority (70–100) of the voice alarm. Select OFF to disable the voice alarm handling of the device.
AUTO EXIT	Select this checkbox if the voice alarm state should be stopped automatically after the alarm signal is stopped/muted (e.g. no alarm request present).
DATE/TIME	Date and time of the PMX-4CR12 system clock.
INTERNAL CLOCK SET	Opens the system clock settings dialog box.
DCF77 SYNCHRONIZATION	Select this option to synchronize the internal clock of the PMX-4CR12 with the DCF77 signal. Please refer to the manual how to connect an external DCF77 receiver.
DCF77 SIGNAL STATUS	Indicates the DCF77 signal strength: – Green: Signal strength OK – Red: Signal strength not OK
DPM 4000 SYNCHRONIZATION	Select this option to synchronize the internal clock of the PMX-4CR12 with a connected DPM 4000 system.
ENABLE SLAVE CLOCK	Select this checkbox if slave clocks are connected to the PMX-4CR12.
CURRENT SLAVE CLOCK TIME	Set the time for the slave clocks.
FIRMWARE	Indicates the firmware version of the PMX-4CR12 when on-line.
UPDATE	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
REBOOT	Reboots the PMX-4CR12.

Security Dialog

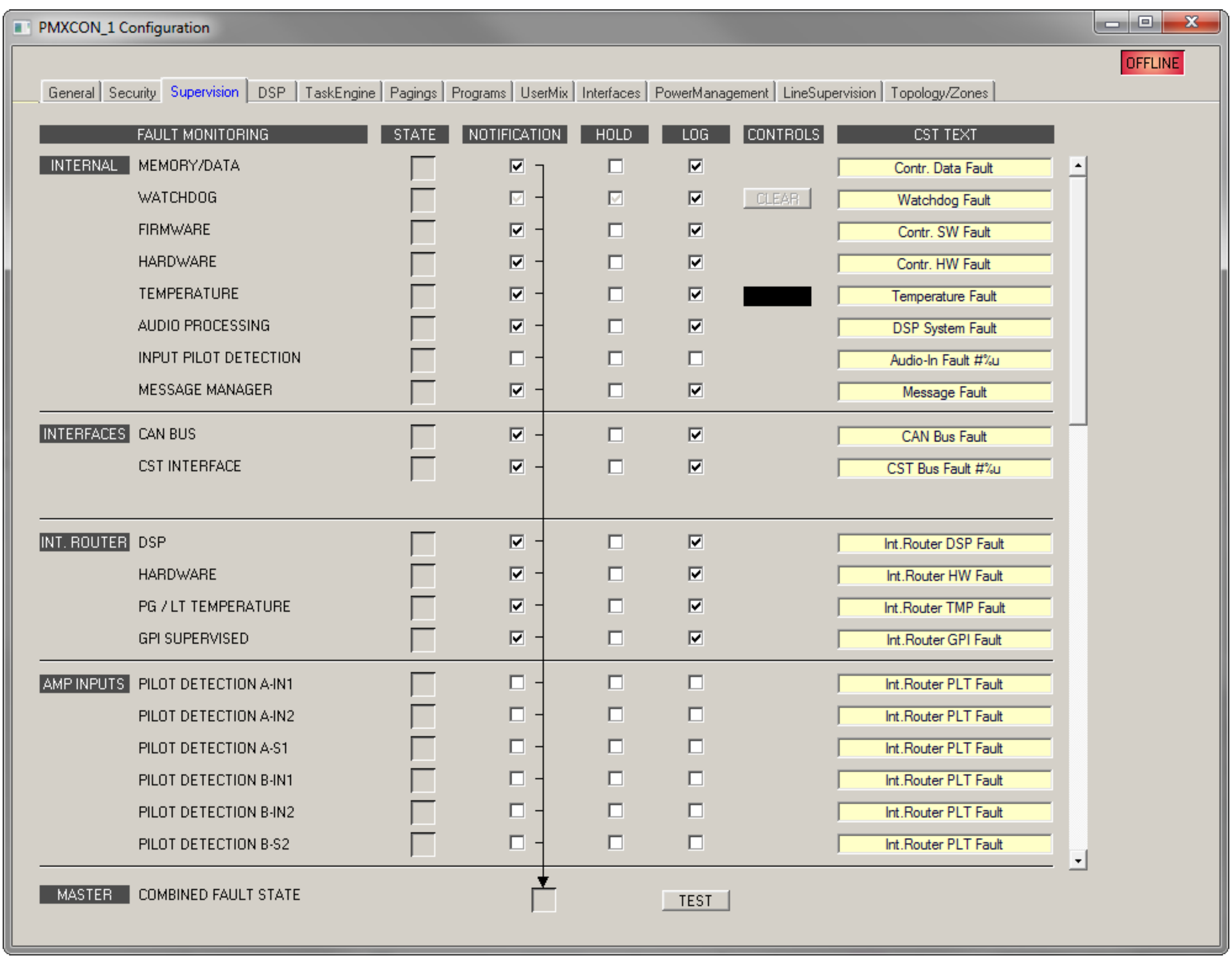
In this dialog the password of the devices can be edited.



Element	Description
LOCK PASSWORD (ACCESS LEVEL 2)	Press the change button to edit the setting of the password for locking call stations.
CST SETUP PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for th setup of call stations.
GO ONLINE PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for going online in IRIS-Net.
FIRMWARE UPDATE PASSWORD (ACCESS LEVEL 4)	Press the change button to edit the setting of the password for updating the firmware of the system.

Supervision Dialog

The Supervision window shows the condition of the PMX-4CR12. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT-LED on the front panel of the device lights, the FAULT relay opens and a signal sound.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the corresponding HOLD checkbox is unchecked.
LOG	
CONTROLS	
CST TEXT	If call stations are configured for error indication, the text entered here is indicated in the call station display if the error occurs. **HINT: The meaning of the parameter %u is described at the error types below.**

INTERNAL

MEMORY/DATA	Memory or Read/Write error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The device firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
HARDWARE	Error in the power supply or the A/D converters of the PMX-4CR12.
TEMPERATURE	Temperature overload of the PMX-4CR12.
Temperature control	Current temperature on the inside of the enclosure.
AUDIO PROCESSING	Error during the processing of audio data.
MESSAGE MANAGER	Error in the message manager.

INTERFACES

CAN BUS	Fault condition on the CAN bus. Further details are provided in the Interface dialog.
CST INTERFACE	Fault condition on the PCA bus. Further details are provided in the Interface dialog. The parameter %u gives the slot number of the erroneous module.

INT. ROUTER

DSP	Error in the digital signal processing (DSP) of the device.
HARDWARE	Hardware error.

PG / LT TEMPERATURE	Temperature overload of the device.
GPI SUPERVISED	Voltage at supervised GPI out of range.

AMP INPUTS

PILOT DETECTION x-IN1	Missing pilot tone at input 1 of cluster A or B.
PILOT DETECTION x-IN2	Missing pilot tone at input 2 of cluster A or B.
PILOT DETECTION A-S1	Missing pilot tone at spare input 1 of cluster A.
PILOT DETECTION B-S2	Missing pilot tone at spare input 2 of cluster B.

EXTERNAL

CALL STATIONs	A connected DPC call station has transferred an error message. The parameter %u gives the address of the erroneous call station.
AMPLIFERS	A connected DPA power amplifier has transferred an error message. The parameter %u gives the address of the erroneous amplifier.
ROUTERS	A connected DCS system has transferred an error message. The parameter %u gives the address of the erroneous DCS system.
POWER SUPPLY	Fault condition in the power supply of the PMX-4CR12.
SPEAKER LINE FAULT	Fault condition in the speaker line supervision. The parameter %u gives the number of the erroneous speaker line, the number has following meaning: 1 to 500: Zone A 501 to 1000: Zone B

USER

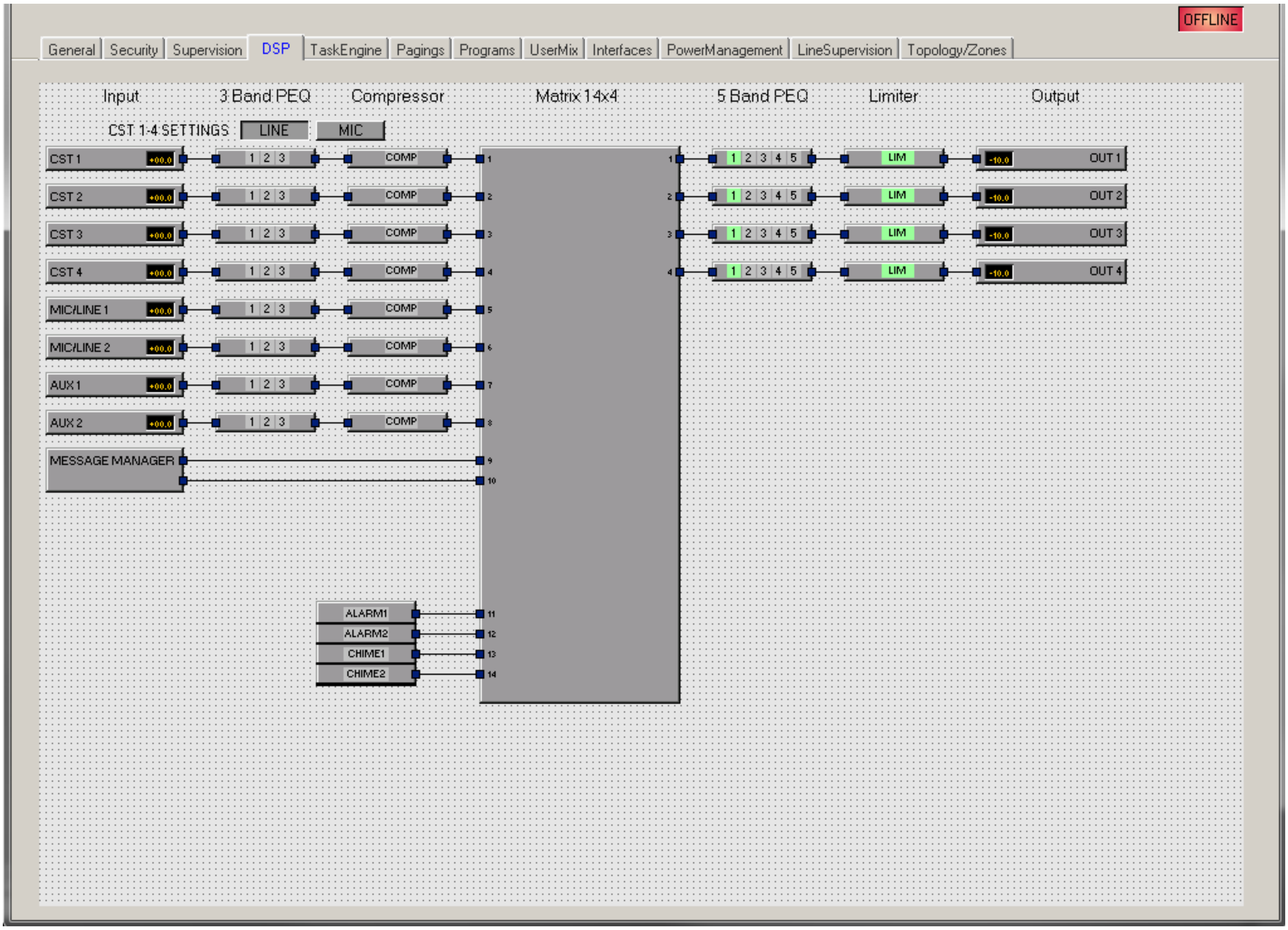
USER FAULT 1 to 10	One or more USER FAULTS have been set **HINT: Use the device Task Engine to configure USER FAULTS.**

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

DSP Dialog

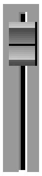
In this dialog the DSP configuration of the Controller is shown. Double clicking on a DSP-Block's icon allows editing its configuration and settings in detail.



Input

The Input block provides access to the audio inputs of the device. The name and gain values of the input channels are indicated in the block. Double click the block to open the Inputs dialog.

Element	Default	Range	Description
CST 1 to 4; MIC/LINE 1,2; AUX 1,2			Permanent channel labeling.
CAN TERM/ STATE			Press the OFF button to activate the internal CAN termination resistor of the corresponding CST bus. The digit next to the button indicates the total number of termination resistors activated. The number must always be “2”.
	0.0 dB	0 to 60 dB	The gain of the MIC/LINE input channels can be adjusted in 6 dB steps.
			The +48V button of the MIC/LINE input channels is for activating phantom power whenever a suitable condenser microphone is being used.

	0.0 dB	-80 to +18.0 dB	Fader for setting the input level.
	0.0 dB	-80 to +18.0 dB	The fader display shows the numerical value of the current fader setting and additionally allows entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) pilot tone detection. The PLT button lights red when pilot tone detection is active but without a pilot signal being detected. With a pilot signal present, the PLT button lights green.
			MUTE button for muting the input signal.
LINE/MIC			Press the LINE button if audio sources with line level signals are connected to the CST busses. Press the MIC button if microphone signals are connected.

MESSAGE MANAGER

The Message Manager block provides access to the messages of the internal Message Manager. Double click the block to open the Message Manager dialog.

Element	**Description**
MESSAGES	
Active	Indicates the messages that are currently active (marked with a “X”).
Description	The unique name or description of the uploaded message. Use the corresponding text field to edit the description. The description can be edited in offline or online mode.
Type	Available message types are EVAC, Chime or Business. The Type can be set when adding messages.
Duration	The duration of the uploaded message, given in format “minutes:seconds”.
Level	Indicates the level of the message. Level ranges from -80 dB to +18 dB. Default level is 0.0 dB. Use the corresponding spin control to edit the level. The level can be edited in offline or online mode.
Info	The memory usage is indicated for all MM-2 modules.
ADD	Press the ADD button to upload a new message. A file selection dialog appears (see screenshot below) that allows selecting a message in WAV file format (mono, 48 kHz). You have to assign a description and a message type (EVAC; Chime or Business) to the message before uploading. If there are two MM-2 modules available, the location for the message has to be selected. **HINT: A selection of standard evacuation messages in different languages is available in the Download area at www.dynacord.com**
DELETE	Press the DELETE button to delete the message selected in the message list.

REPLACE	Press the REPLACE button to replace the message selected in the message list, the message type and location can not be changed. In online mode only business messages can be replaced.
ERROR STATES	
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR flag is set at the same time. Additionally the FAULT-LED on the front panel of the DEVICE lights, the FAULT relay opens and a signal sounds.
MODULE	Hardware or configuration error in the MM-2 module.
MESSAGE STORAGE	Error during message storage.
PLAYBACK MEMORY	Error in the playback memory.
WATCHDOG	Watchdog error of the DEVICE. This error type is logged conforming to standards.
TEMPERATURE	Temperature of module is too high.
COLLECTED ERROR	The FAULT-LED on the front panel of the PMX-4CR12 lights at the occurrence of this type of error.
FALLBACK SIGNALS	
Fallback Evac	Select the default evacuation signal to use if no message is uploaded to the MM-2 module. This settings is valid for all PMX-4CR12 in the PROMATRIX system.
Fallback Pre-/ Chime	Select the default chime or pre-chime signal to use if no chime is uploaded to the MM-2 module. This settings is valid for all PMX-4CR12 in the PROMATRIX system.

HINT: For creating audio messages the software Audacity from http://audacity.sourceforge.net/ **can be used.**

3 BAND PEQ
Equalizers accentuate or lower the audio signal within specific frequency ranges. Eight parametric 3-Band equalizers are available.

Element	Default	Range	Description
CST 1-4, MIC/LINE 1-2, AUX 1-2			Press the input channel button to view or edit the corresponding PEQ settings.
LINE/MIC			Press the LINE button if audio sources with line level signals are connected to the CST busses. Press the MIC button if microphone signals are connected.
BYPASS ALL			Pressing BYPASS ALL switches off all filters.

EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope.
GAIN +0.0 dB	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
FREQ 30.0 Hz	125 Hz, 1 kHz, 16 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.1 to 100 (PEQ) 0.1 to 2.0 (Hi-/ Lopass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi and Lo pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

COMPRESSOR

The compressor reduces the dynamic range of audio signals. Once the signal exceeds a certain threshold, the signal gets compressed, i.e. major input level changes result in minor output level changes. Narrowing the dynamic range often allows for easier recording or mixing the audio signal. Eight compressors are available.

Element	Default	Range	Description
CST 1-4, MIC/LINE 1-2, AUX 1-2			Press the input channel button to view or edit the corresponding Compressor settings.

THRESHOLD +0.0 dBu 0.775 V	+6.0 dBu or 1.546 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	THRESHOLD defines the signal level at which the Compressor sets in. Entering the desired value is possible in dBu as well as in V. The entered value is automatically converted in both directions.
RATIO 1.0:1	4.0:1	1.0:1 to 8.0:1	RATIO defines the compression rate, i.e. the degree of compression above the threshold level. For example, a rate of 4.0 : 1 represents a signal reduction by factor 4.
ATTACK 5 ms	5 ms	0 to 99 ms	ATTACK defines the velocity, at which the compressor sets in. A short attack rate means that even short signal peaks are efficiently com- pressed. Longer attack rate leave signal peaks untouched.
RELEASE 250 ms	250 ms	0 to 999 ms	RELEASE defines the control time interval the compressor takes to return to an uncompressed signal level, after the signal dropped below the set threshold.
BYPASS			BYPASS activates (not engaged) or deactivates the Compressor (engaged), which allows for quick A / B comparison between the com- pressed and uncompressed audio signal.

5 BAND PEQ

Equalizers accentuate or lower the audio signal within specific frequency ranges. Four parametric 5-Band equalizers are available.

Element	**Default**	**Range**	**Description**
OUT 1-4			Press the output channel button to view or edit the corresponding Parametric EQ settings.
BYPASS ALL			Pressing BYPASS ALL switches of all filters.
EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope.

GAIN +0.0 dB	0 dB	-18 to +18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
FREQ 30.0 Hz	60 Hz, 250 Hz, 1 kHz, 4 kHz, 19 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.01 to 6.67 Oct. or 0.1 to 40 (PEQ) 0.1 to 2.0 (Hi-/Lopass)	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi and Lo pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

LIMITER

A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage.

Element	**Default**	**Range**	**Description**
OUT 1-4			Press the output channel button to view or edit the corresponding Peak Limiter settings.
THRESHOLD +0.0 dBu 0.775 V	+6.0 dBu or 1.546 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter unaffected. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Entering the threshold value is possible in dBu or V. The value can be entered in either box and will automatically be converted in the other.
ATTACK 5 ms	5 ms	0 to 50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.

	100 ms	10 to 1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

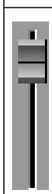
OUTPUT

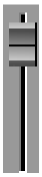
The Output block provides access to the outputs of the device. The name and gain values of the out channels are indicated in the block. Double click the block to open the Output dialog.

Element	Default	Range	Description
OUT 1-4			Permanent channel labeling.
	0.0 dB	-80 to +18.0 dB	Fader for setting the output level.
	-10.0 dB	-60 to +6 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) the pilot tone generator.
			MUTE button for muting the output signal.

ALARM CHIME

The Alarm Chime dialog allows the configuration of the internal alarm and chime generators.

Element	Default	Range	Description
Alarm Configuration			
	-3.0 dB	-80 to 0 dB	Fader for setting the alarm level.
	-3.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
NEW ALARM			Press this button to ad a new alarm to the alarm list.
PLAY ALARM			Press this button to playback the alarm selected in the alarm list.

Chime Configuration			
	-9.0 dB	-80 to 0 dB	Fader for setting the chime level.
	-9.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
PLAY CHIME			Press this button to playback the chime selected in the chime list.

MATRIX

Double click on the Matrix 14x4 to open the Matrix 10x4 dialog (the 4 missing inputs in this dialog are used for the internal generators of the PMX-4CR12). The Matrix 10x4 allows connecting inputs and outputs. Left clicking the node in the matrix where the output channel's column and the input channel's line meet with the mouse does connect an output to an input. Clicking again onto the corresponding node disconnects inputs and outputs.
Please note following restrictions for making connections in the matrix:
- BGM inputs can only routed via a call station, so this is not possible in this dialog
- Unused inputs can not be routed
- Inputs used for alarms, announcements etc. can not be routed
- Inputs used for the Message Manager can not be routed
- Manual routings override existing BGM routings

Element	**Default**	**Range**	**Description**
DUCKING	-40 dB	-85 to 0 dB	The signal level of the back ground music is reduced by the level entered here when the in input signal, signal level reaches or exceeds a set threshold.
FADE IN	0.02 s	0.01 to 4 s	FADE IN defines how fast the gain of the background music signal is reduced after the input signal exceeds the threshold level.
FADE OUT	0.02 s	0.01 to 0.4 s	FADE OUT defines how fast the gain of the background music signal is returned to the preset level once the input signal drops below the threshold level.

TaskEngine Dialog

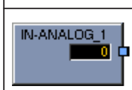
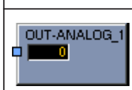
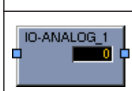
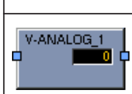
The Task Engine Window allows configuring the Task Engine by dragging inputs, links or outputs from the categories of the FUNCTIONS AND IOS on the left corner of the screen into the Task Engine Worksheet. Elements can be freely positioned and wired within the worksheet. Double clicking on inputs or outputs allows configuring them in detail. Copy & paste of blocks allow convenient editing of the Task Engine configurations. The size of the worksheet automatically increases when a block is moved to the current border.
Configuring the Task Engine as well as wiring DSP blocks is possible only in offline mode. Please refer to section "How to configure a Control" on page 20 how to assign functions and connections to a Task Engine block.

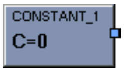
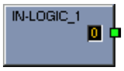
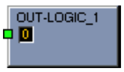
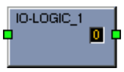
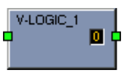
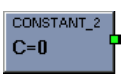
In the Task Engine, one differentiates between two classes of variables:

– Analog: variables of the type "analog" are rational numbers. Example: Level value (-80...+18) of a DSP block mono mixer output.
– Logic: variables of the type "logic" are Boolean values, i.e. only the values "0" and "1" are allowed. Example: Mute (0 = not muted, 1 = muted) of a DSP block monaural mixer output.

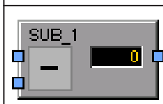
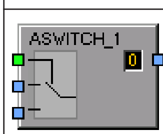
In the Task Engine, different colors are used to distinguish the two types of variables. Inputs and outputs that are not wired are marked blue, whenever variables of the type "analog" are being processed or transmitted. Inputs and outputs that are not wired are marked green, whenever variables of the type "logic" are being processed or transmitted.

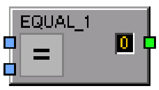
VALUES

Element	Description
	The Input Analog block is a variable parameter for rational numbers which always outputs the current value of the connection.
	The Output Analog block is a variable parameter for rational numbers which always assigns the current value at the input to the connection.
	The IO Analog block is a variable parameter for rational numbers which always outputs the current value of the connection. The current value at the input is assigned to the connection.
	The Virtual Analog Value block is similar to the IO Analog block, but has no connection. The keyword Value of the block is used instead of a connection. The keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the DPM. – Persistent = 0: The value is stored in volatile memory.

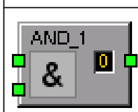
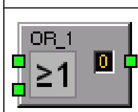
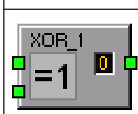
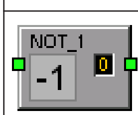
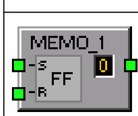
	The Constant Analog block is a constant parameter for rational numbers. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Input Logic is a variable parameter for Boolean values which always outputs the current value of the connection.
	The Output Logic block is a variable parameter for Boolean values which always assigns the current value at the input to the connection.
	The IO Logic block is a variable parameter for Boolean values which always outputs the current value of the connection. The current value at the input is assigned to the connection.
	The Virtual Logic block is similar to the IO Logic block, but has no connection. The keyword Value of the block is used instead of a connection. he keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the device. – Persistent = 0: The value is stored in volatile memory.
	The Constant Logic block is a constant parameter for Boolean values. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Calendar Entry block is being used to create time-dependant Boolean values. The outputted Boolean value depends on the configuration of this block and the current system time.

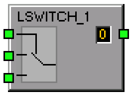
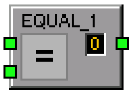
ANALOG OPERATIONS

Element	Description
	The Addition block provides 2 inputs for rational numbers. The rational number at the output is always the sum of rational numbers of the (wired) inputs.
	The Subtraction block subtracts the rational number of the lower input from the rational number of the upper input. The output always presents the result of this analog operation.
	The Multiplication block multiplies the rational number of the upper input with the rational number of the lower input. The output always presents the result of this analog operation.
	The Division block divides the rational number of the upper input by the rational number of the lower input. **CAUTION: If the rational number "0" is present at the lower input, the rational number "0" is always output, independent of the upper input's value.**
	The Switch block switches the rational number at the center or lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.

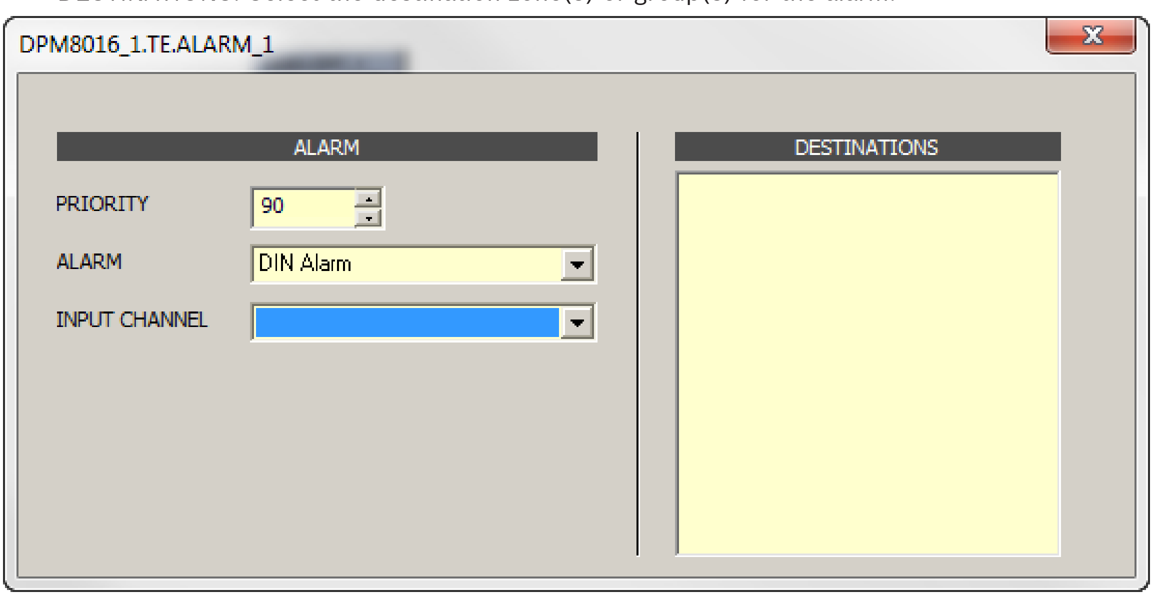
	The Convert block converts a Boolean value to a rational number. The Boolean value 0 is converted to the rational number 0.0, the Boolean value 1 is converted to the rational number 1.0.
	The Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when identical numbers are present at the inputs.
	The Not Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the numbers that are present at the inputs differ from each other.
	The Greater block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than the rational number of the lower input.
	The Greater Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater or equal than the rational number of the lower input.
	The Less block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower than the rational number of the lower input.
	The Less Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower or equal than the rational number of the lower input.
	The Range Check block allows the range check of a rational number. The Boolean value at the output is only true when the rational number at the input is greater or equal than the property TE.RANGE_CHECK_1.range.min **and** lower or equal than the property TE.RANGE_CHECK_1.range.max – range.max: Enter the upper bound of the range to be checked – range.min: Enter the lower bound of the range to be checked

LOGIC OPERATIONS

Element	Description
	The AND block provides 2 inputs for Boolean values. The Boolean value at the output is only true when all (wired) inputs are true.
	The OR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when at least one (wired) input is true.
	The XOR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when exactly one (wired) input is true.
	The NOT block negates the Boolean value at the input.
	The Memo (Flip-flop) block provides 2 inputs for Boolean values. The upper input sets the flip flop, the lower input resets the flip flop.

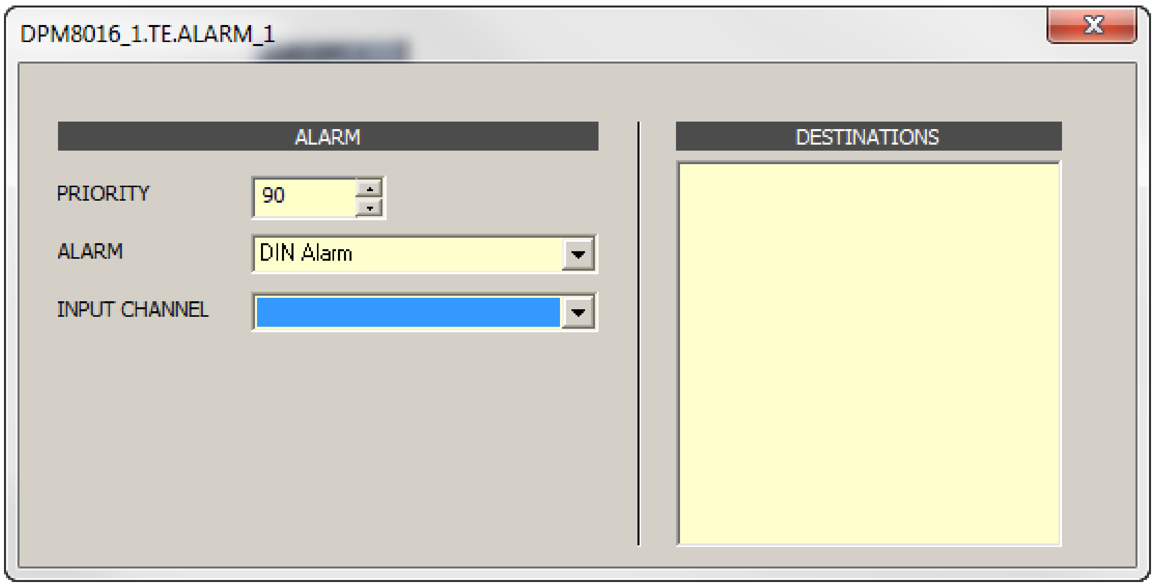
	The Switch block switches the Boolean value at the center or the lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Convert block converts a rational number to a Boolean value. The rational number 0.0 is converted to the Boolean value 0, the rational number 1.0 is converted to the Boolean value 1.
	The Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are identical (e.g. both inputs are true, or both inputs are false).
	The Not Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are different from each other (e.g. one input is true while the other input is false).

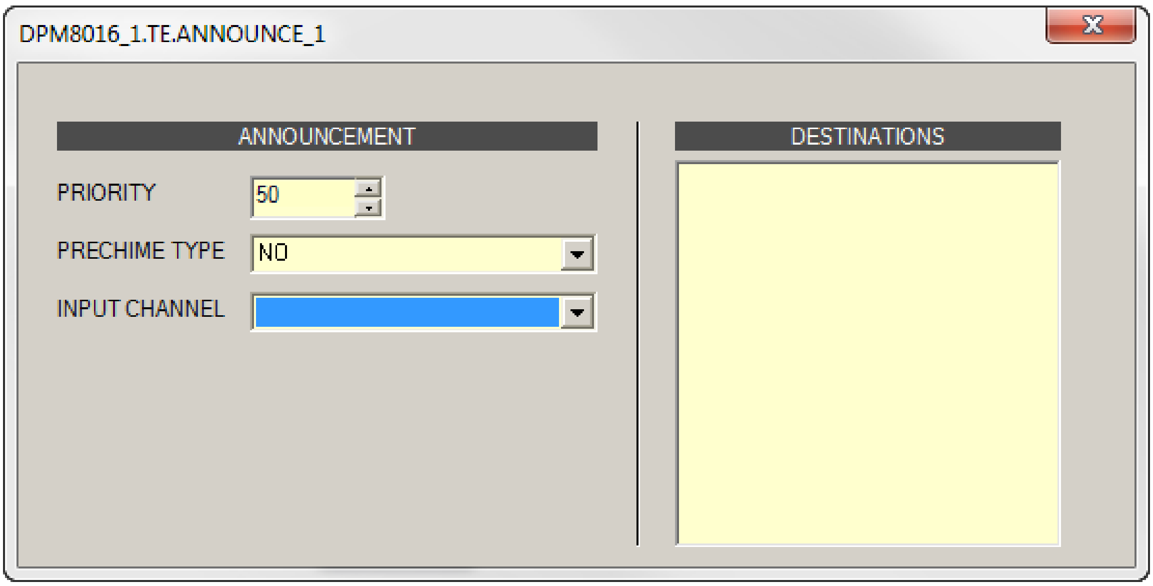
ADVANCED OPERATIONS

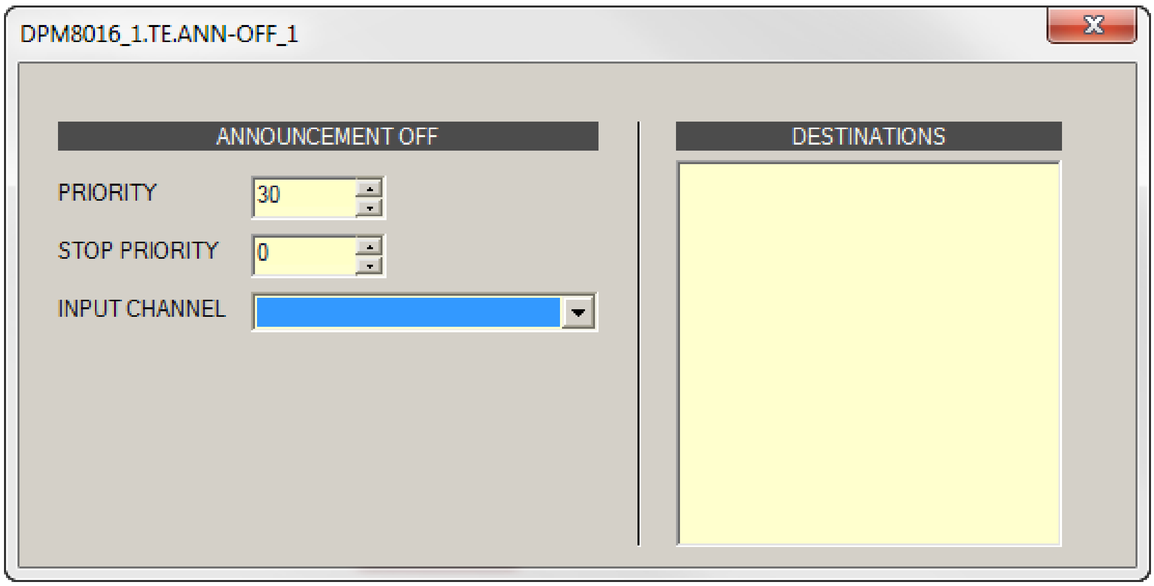
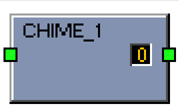
Element	**Description**
	The Alarm block is used to trigger an alarm. Double click the block to edit the alarm settings (see screenshot below). – PRIORITY: Priority of the alarm (0 to 100). – ALARM: Select the alarm type to be triggered, see table below. – INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the DPM 8016, where the external alarm signal is present. – DESTINATIONS: Select the destination zone(s) or group(s) for the alarm. 

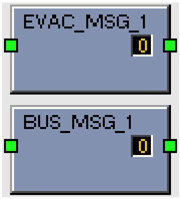
The Manual Alarm is similar to the Alarm block. The additional input T acts like a pushbutton and allows to switch the alarm signal on or off. Double click the block to edit the alarm settings (see screenshot below).

– PRIORITY: Priority of the alarm (0 to 100).
– ALARM: Select the alarm type to be triggered, see table below.
– INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the device, where the external alarm signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s) for the alarm.



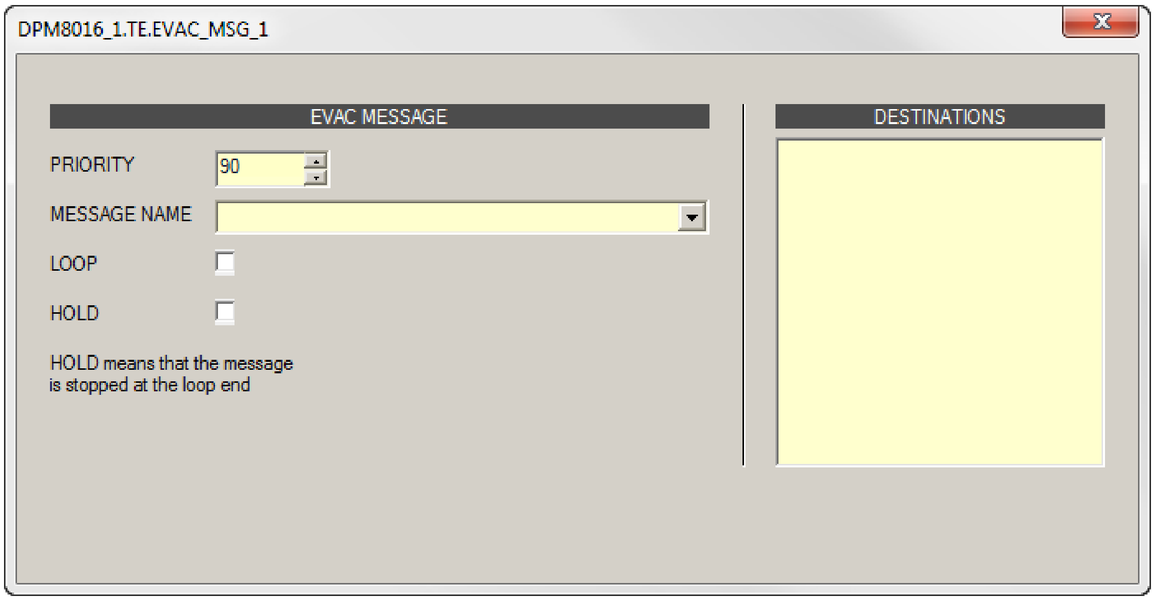
<table>
<tr>
<td>
</td>
<td>The Announcement block is used to trigger an announcement. Double click the block to edit the announcement settings (see screenshot below).
– PRIORITY: Priority of the announcement (0 to 100).
– PRECHIME TYPE: Select the pre chime (see table below). Select NO, if there should be now pre chime
– INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s) for the announcement.

</td>
</tr>
</table>

<table>
<tr>
<td>
</td>
<td>The Announcement OFF block is used to stop an announcement. Double click the block to edit the announcement settings (see screenshot below).
– PRIORITY: Priority of the announcement (0 to 100).
– STOP PRIORITY: Enter the priority (0 to 100) that is used to stop an announcement.
– INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s), where the announcement should be stopped.

</td>
</tr>
<tr>
<td>
</td>
<td>The Chime block is used to trigger a chime. Double click the block to edit the chime settings.
– PRIORITY: The priority of the chime (0 to 100).
– TYPE: Select the type of the chime.
– LOOP: Select this checkbox to repeat the chime automatically.
– HOLD: Hold means that the message is stopped at the loop end.
– DESTINATIONS: Select the destination zone(s) or group(s) for the chime.</td>
</tr>
</table>



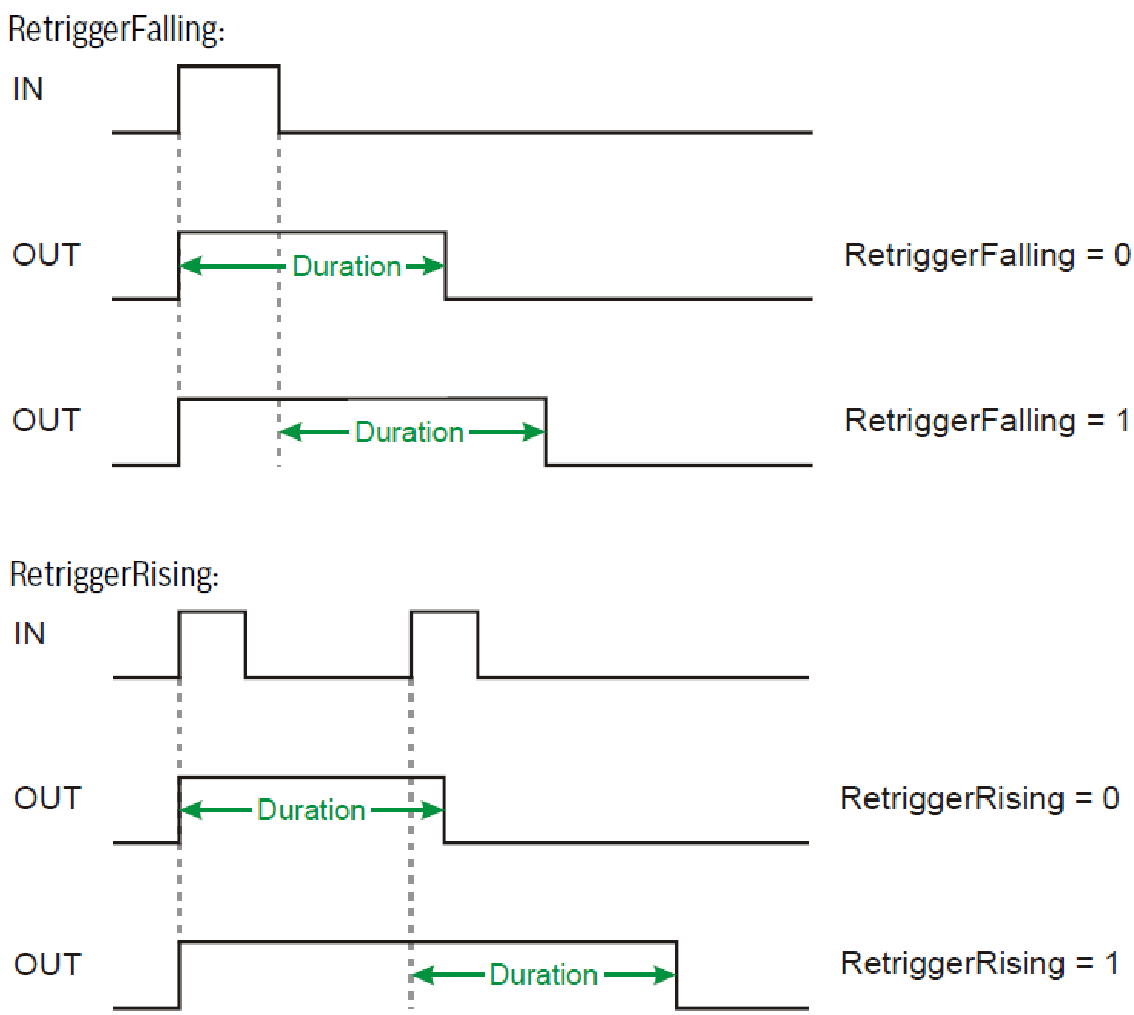
The EVAC Message or Business Message block is used to trigger a MM-2 message. Double click the block to edit the message settings (see screenshot below).

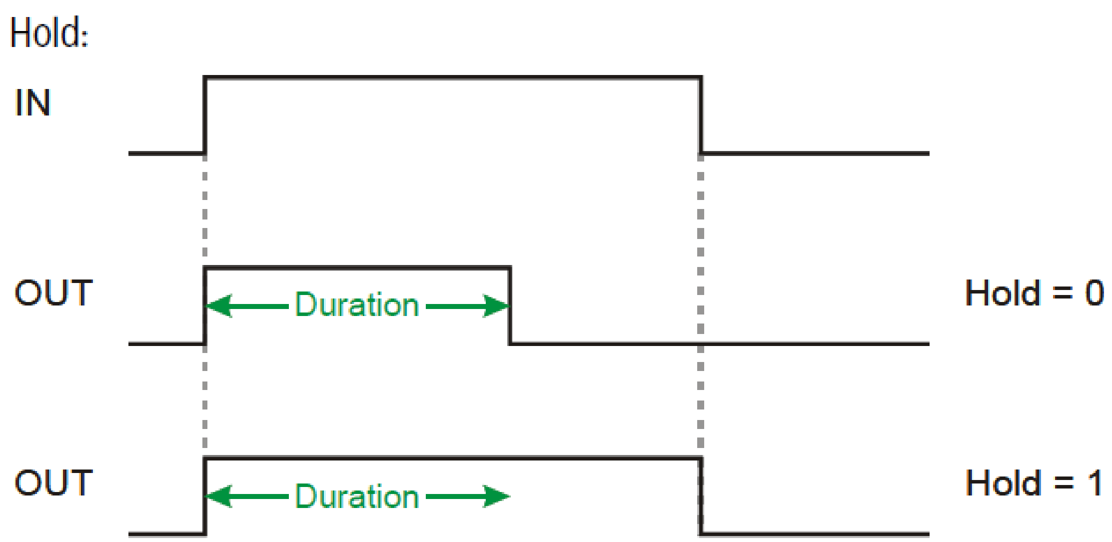
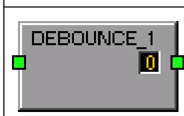
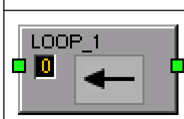
– PRIORITY: The priority of the message (0 to 100).
– MESSAGE NAME: Select the (EVAC or Business) message to start.
– LOOP: Select this checkbox to repeat the message automatically.
– HOLD: Hold means that the message is stopped at the loop end.
– DESTINATIONS: Select the destination zone(s) or group(s) for the message.



The Timer block sets the state at the output to true for an adjustable duration, when the Boolean value at the input changes from false to true.

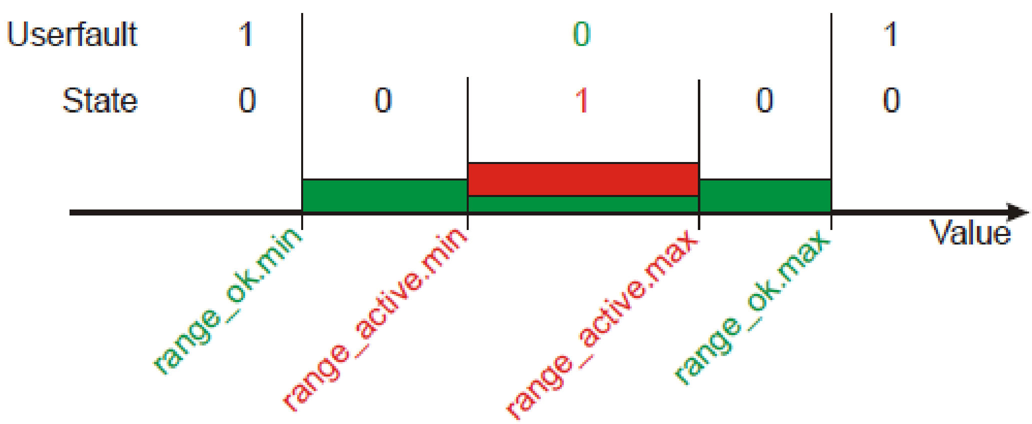
– Duration: Enter the duration in seconds, without unit.
– Hold: See illustration below.
– Retrigger Falling: See illustration below.
– Retrigger Rising: See illustration below.
– State: State of the block (1 = time running)
– Timer Value



	
	The CST Text block is used for indicating a text message at the LC display of one or more call stations. – Acknowledge: Enter 1 if pressing the ESC button at the call station should discard the text at the display. – Address: Enter the CAN address of the call station, where the text should be indicated. Enter 0 if the text should be indicated at all call stations. – Buzzer: Enter 1 if the buzzer should signal the text indication additionally. – Clear: Enter 1 if the text should be cleared when the input changes from true to false. – Duration: Enter the time in seconds (without unit), how long the text should be indicated. – State: State of the block (1 = text is indicated) – Text: Enter the text to be indicated at the display. The maximum length is 20 characters, including space and special characters. See table below for available characters.
	The Impedance Measurement block is used for executing a line measurement. – Lines By Name = ALL – State: State of the block (1 = measurement active) – Test Function = LINETEST
	The Debounce block is used for debouncing a signal. – Falling Edge: Enter 1 if the falling edge (change from true to false) at the input should be debounced. – Rising Edge: Enter 1, if the rising edge (change from false to true) at the input should be debounced. – State: State of the block – Time: Enter the debounce time in seconds (without unit)
	The Loop block allows building feedback loops in the Task Engine. Unstable conditions are prevented by the block. To point out the function of this block, the input is at the right side, the output is at the left side.
Text Box	The Text Box allows labeling the task engine configuration. Click the Modify Properties entry in the context menu to open the Edit Textbox dialog. This dialog allows editing the caption and e.g. font size and font type.

<table>
<tr><td></td><td>The Input Supervision block allows supervision of a rational number, especially an input signal from a CIE (Control and Indicating Equipment/fire alarm system). Two ranges can be defined, the Active range and the Ok range. Depending on the ranges the Boolean value at the output (e.g. for triggering an alarm) and a USER FAULT (e.g. for error indication of invalid input values) will be set.
The Active range is defined by:
– range_active.max: Upper bound of the Active range
– range_active.min: Lower bound of the Active range
The Boolean value at the output is true if the rational number assigned via Function & Connection is within the Active range. The Boolean value at the output is false if the rational number at the input is below or above the Active range.
The Ok range is defined by:
– range_ok.max: Upper bound of the Ok range.
– range_ok.min: Lower bound of the Ok range.

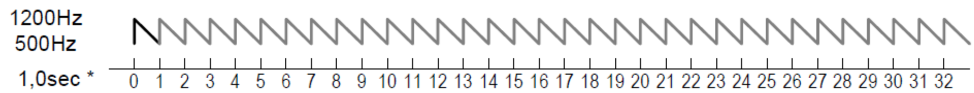
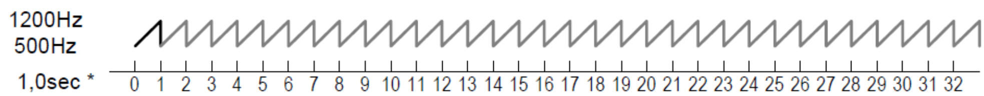
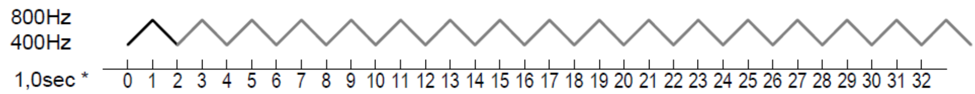
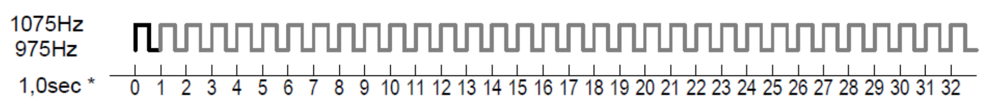
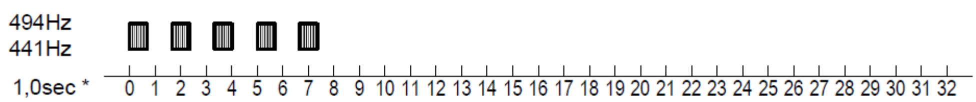
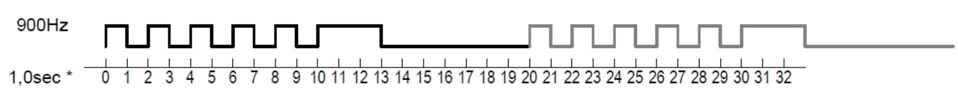
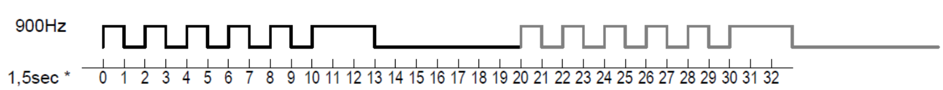
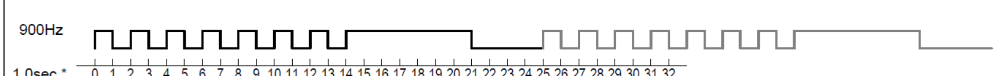
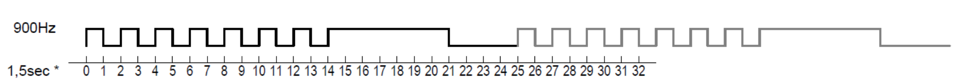
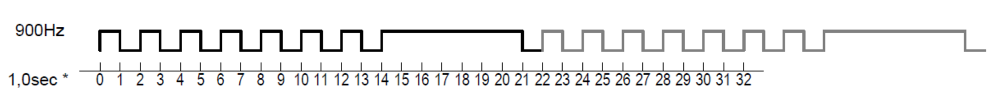
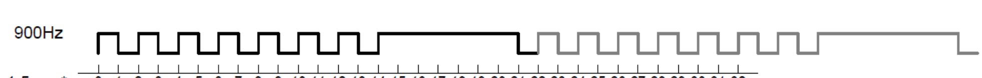
HINT: If the value of the assigned Function & Connection leaves the Ok range, the State does not change (“state value is latched“)

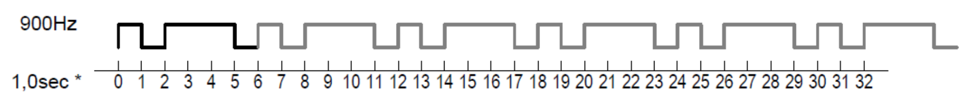
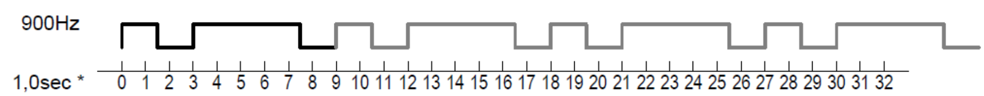
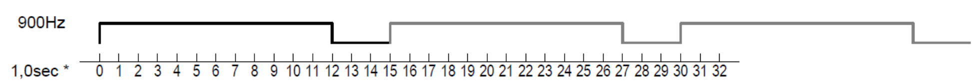
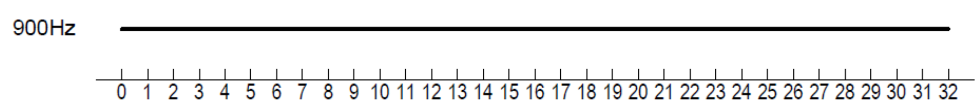
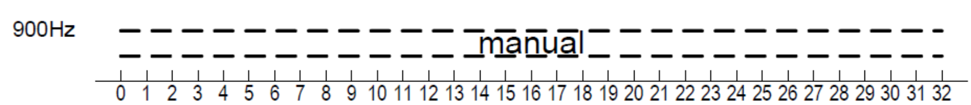
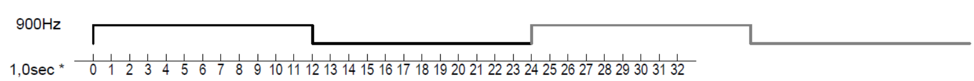
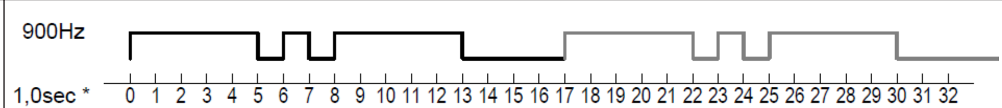
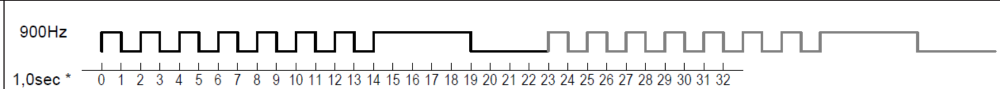
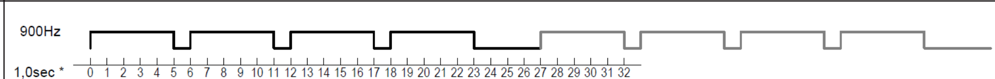
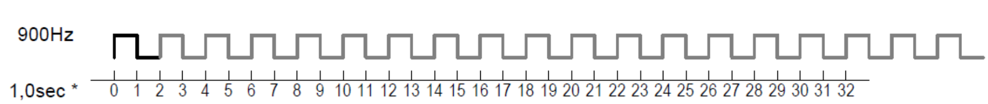
The USER FAULT is set to 0 if the rational number assigned via Function & Connection is within the Ok range. The USER FAULT is set to 1 if the rational number at the input is below or above the Ok range. Following properties are used to select the USER FAULT:
– userfault_connection
– userfault_idx
</td></tr>
<tr><td>Superblocks</td><td>Superblocks are listed here. Please refer to page 240 how to use Superblocks.</td></tr>
</table>

Chime types

Typ
1_TONE
2_TONE
3_TONE
4_TONE
2x2_TONE
2_TONE_PRE

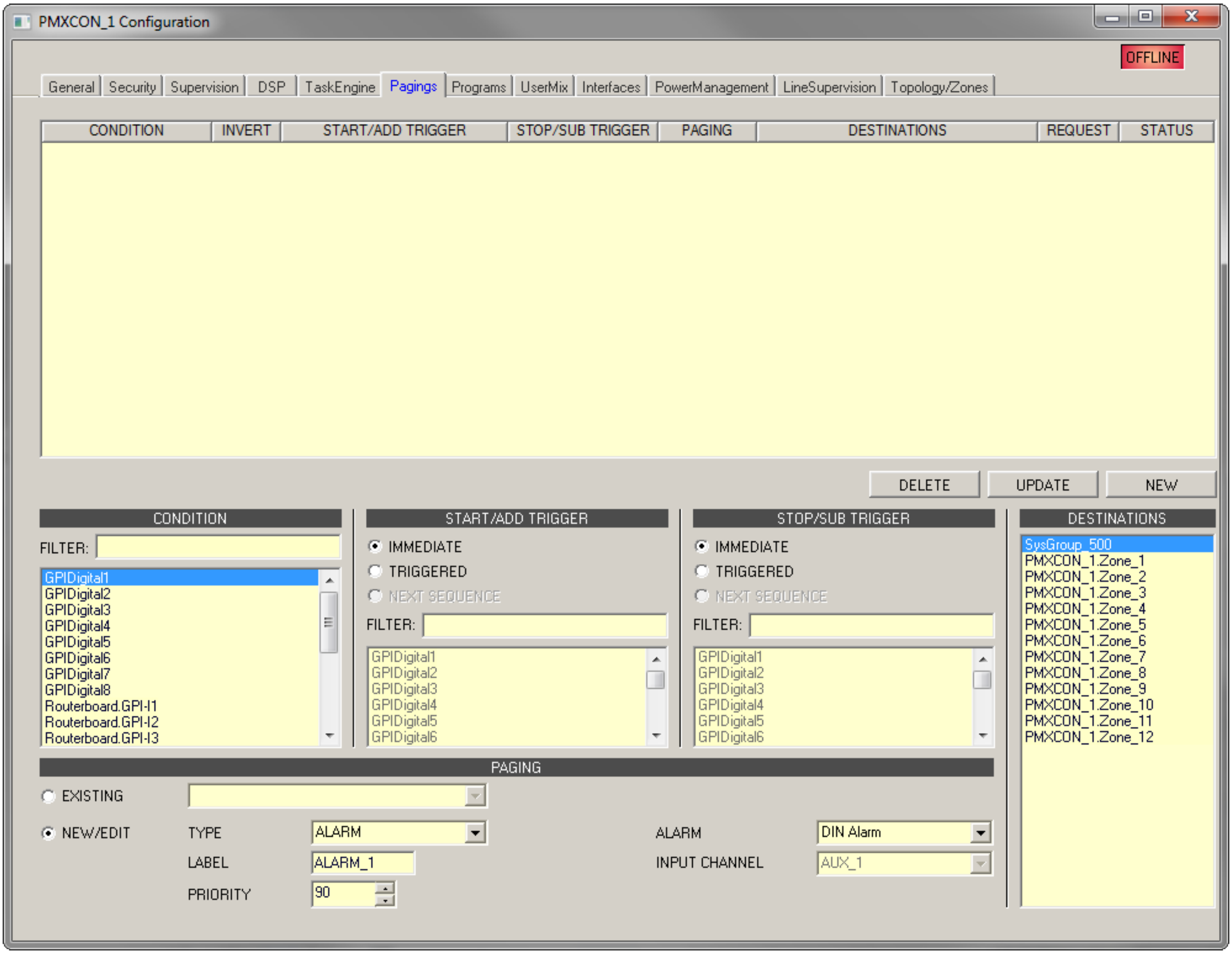
Alarm Types

Type	**Graphical Illustration**
Extern	-
DIN Alarm	
Slow Whoop	
Siren	
Two-Tone Alarm	
Telephone Alarm	
Ship Alarm 1	
Ship Alarm 2	
Ship Alarm 3	
Ship Alarm 4	
Ship Alarm 5	
Ship Alarm 6	
Ship Alarm 7	

Ship Alarm 8	
Ship Alarm 9	
Ship Alarm 10	
Ship Alarm 11	
Ship Alarm 12	
Ship Alarm 13	
Ship Alarm 14	
Ship Alarm 15	
Ship Alarm 16	
Ship Alarm 17	
SILENCE	-

Pagings Dialog

This dialog allows the configuration of pagings (e.g. alarm or EVAC messags) with dynamic destinations.



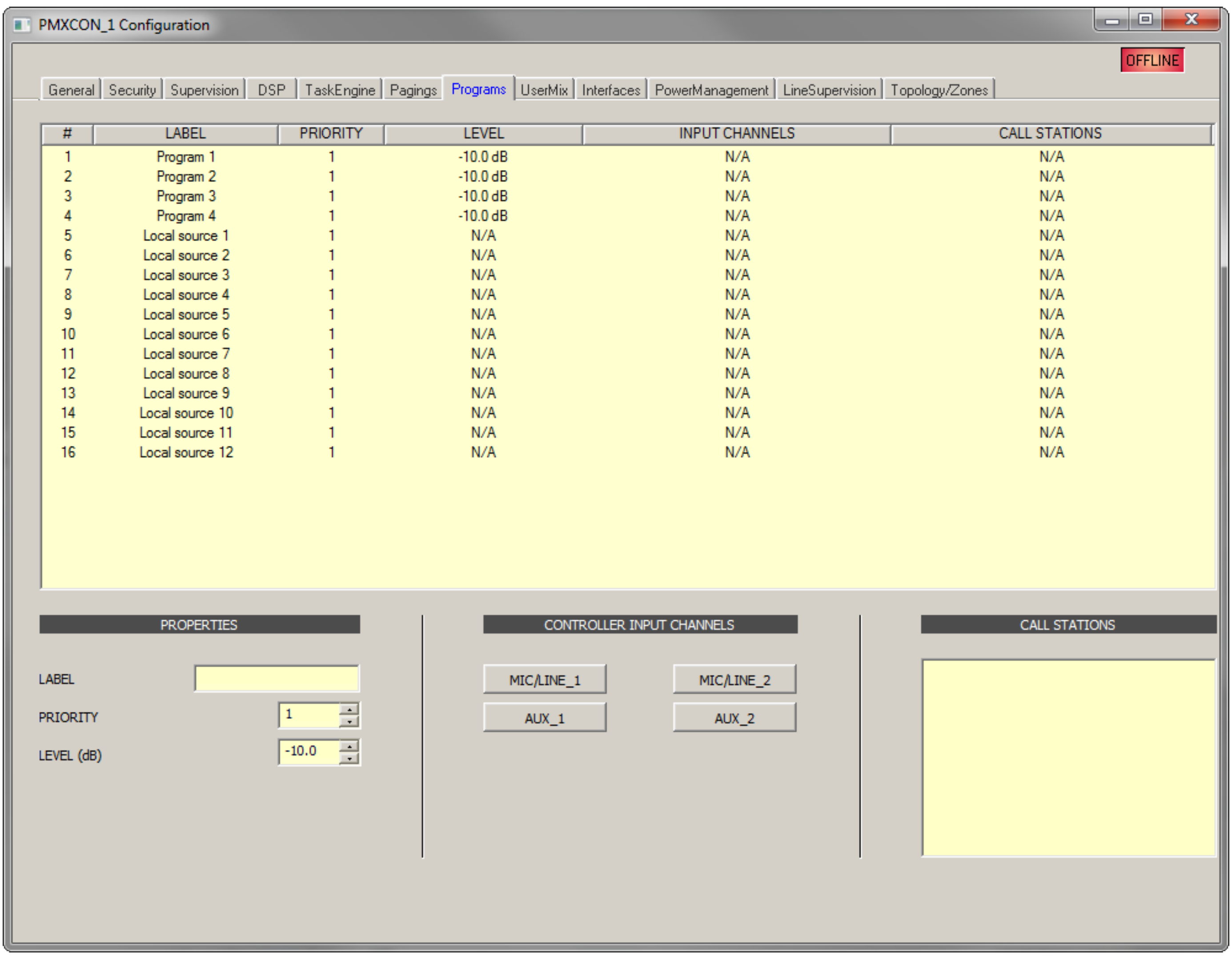
Element	Description
CONDITION	The status of the condition selected here starts the paging, e.g. the contact of a CIE connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the paging.
START/ADD TRIGGER	The value used to trigger the start of (or addition of destinations to) an active paging. The rising edge of the value is evaluated.
STOP/SUB TRIGGER	The value used to trigger the stop of (or subtraction of destinations from) an active paging. The rising edge of the value is evaluated.
PAGING	The paging initiated by the condition.
DESTINATIONS	The destinations (zones or groups) of the paging.
REQUEST	Indicates if the paging condition is active or inactive.
STATUS	Indicates if the paging is ON or OFF.

Element	Description
DELETE	Press the DELETE button to delete the paging selected in the paging list.
UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to paging selected in the paging list.
NEW	Press the NEW button to create a new paging using the settings in the lower section of the dialog and adds it to the paging list.
CONDITION	
FILTER and condition list	Select the condition to start a paging from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
START/ADD TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should start immediately or the zones should be added immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be added only after the message ended. When selected, the paging is started immediately. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
STOP/SUB TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should stop immediately or the zones should be removed immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be removed only after the message ended. When selected, the paging is stopped immediately after the message ended. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
PAGING	
EXISTING	Select EXISTING to select an existing paging from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the paging.
TYPE	Select the paging type from the dropdown.
LABEL	Enter the name of the paging.
PRIORITY	Select the priority of the paging.
ALARM	If the selected paging TYPE = ALARM you can select the alarm type from this dropdown.
PRECHIME TYPE	If the selected paging TYPE = ANNOUNCEMENT you can select the prechime type from this dropdown.

CHIME TYPE	If the selected paging TYPE = CHIME you can select the chime type from this dropdown.
MESSAGE NR	If the selected paging TYPE = EVAC Message you can select the message number from this dropdown.
INPUT CHANNEL	If the selected paging TYPE= ANNOUNCEMENT or TYPE = ALARM (and ALARM = Extern) you can select the audio input channel for the paging.
DESTINATIONS	Select the zones or groups of the paging.

Programs Dialog

The Programs dialog allows to configure 4 programs for back ground music.

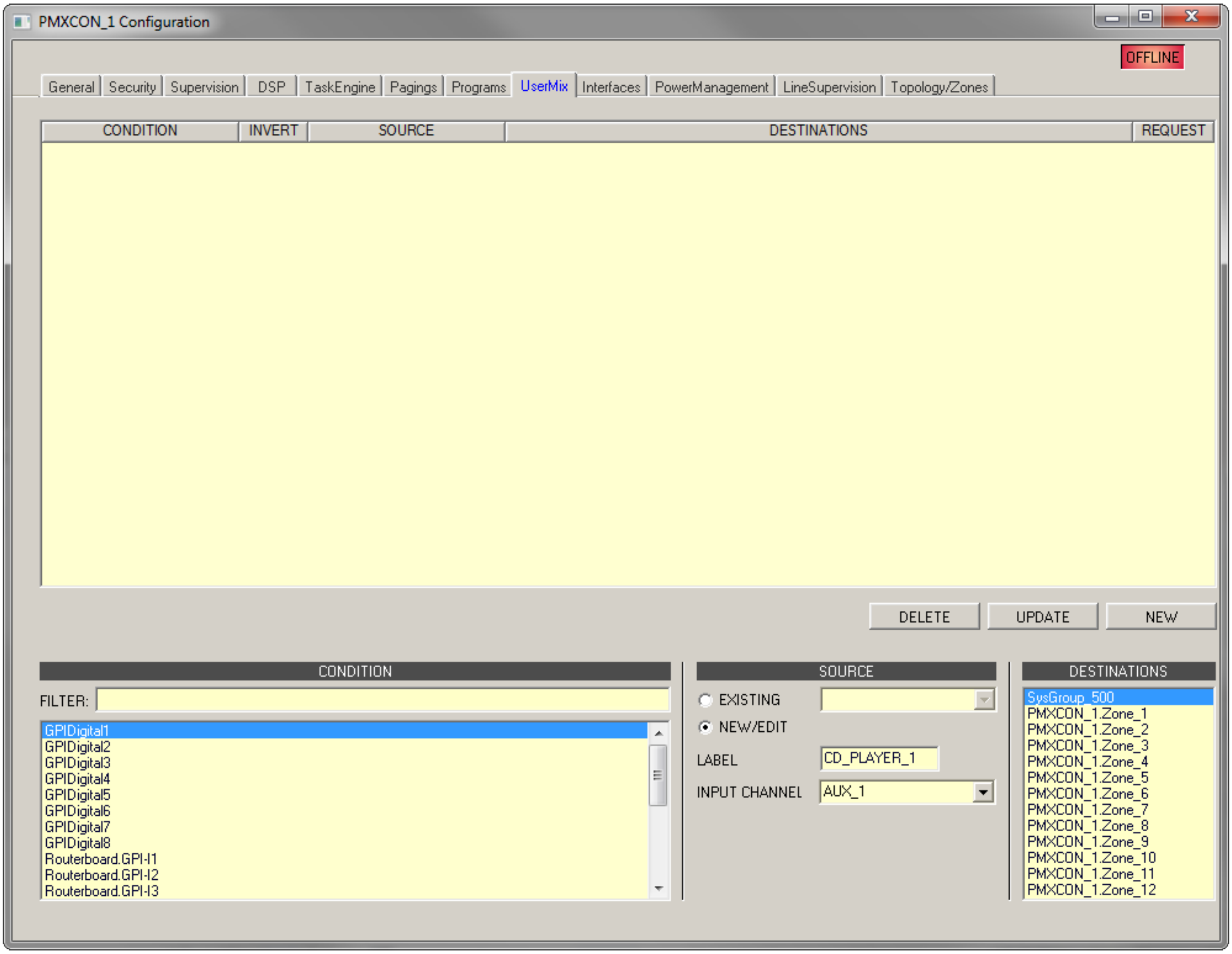


Element	Description
#	Number of the program.
LABEL	Name of the program.
PRIORITY	The priority of the program.

Element	Description
LEVEL	Level of the program.
INPUT CHANNELS	Input channel of the program. Select more than one input channel to mix the audio signals.
CALL STATIONS	The call stations where this program is listed in the menu and can be selected by the call station user.
LABEL	Text field for labeling a program (max. 20 characters), e.g. giving it an application specific name. Note: Using “,” (comma) in a name is not permissible.
PRIORITY	Edit the priority of the program selected in the program list (range: 1 to 69).
LEVEL (dB)	Edit the level of the program selected in the program list (range: -80 to 0 dB). Only the level can be edited in online mode.
CONTROLLER INPUT CHANNEL: MIC/LINE 1-2, AUX 1-2	Select the controller input channel to be used as audio source of the selected program.
AMPLIFIER INPUT CHANNEL	Select the amplifier input channel to be used as local audio source.
CALL STATIONS	Select the call stations where the selected program will be listed in the menu.

UserMix Dialog

This dialog allows configuring audio routings (e.g. background music) in the system.



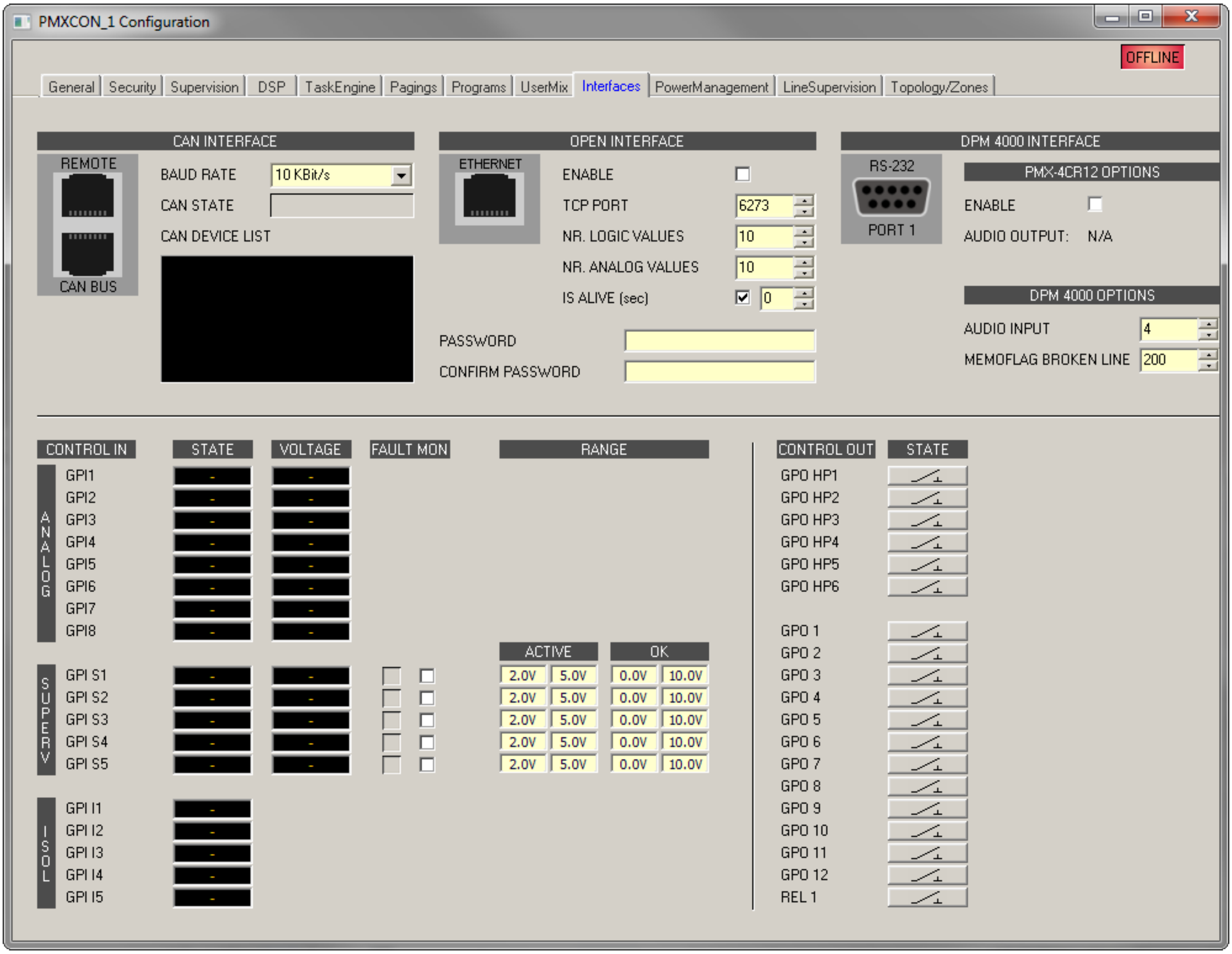
Element	**Description**
CONDITION	The condition that starts the background music, e.g. a switch connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the back ground music.
SOURCE	The source of the background music.
DESTINATIONS	The destinations (zones or groups) of the background music.
REQUEST	Indicates the current status (active or inactive)

Element	**Description**
DELETE	Press the DELETE button to delete the entry selected in the list.
UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to entry selected in the list.
NEW	Press the NEW button to create a new background music using the settings in the lower section of the dialog and adds it to the list.

Element	Description
CONDITION	
FILTER and condition list	Select the condition to start background music from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
SOURCE	
EXISTING	Select EXISTING to select an existing source for background music from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the source.
LABEL	Enter the name of the background music.
INPUT CHANNEL	Select the audio input channel for the background music.
DESTINATIONS	Select the zones or groups of the background music.

Interfaces Dialog

The Interface window allows configuring the different interfaces located on the rear panel of the device. All REMOTE CAN BUS and CONTROL PORT settings can be made in here. Configuring the Ethernet interface is done under Network Settings in the General window.

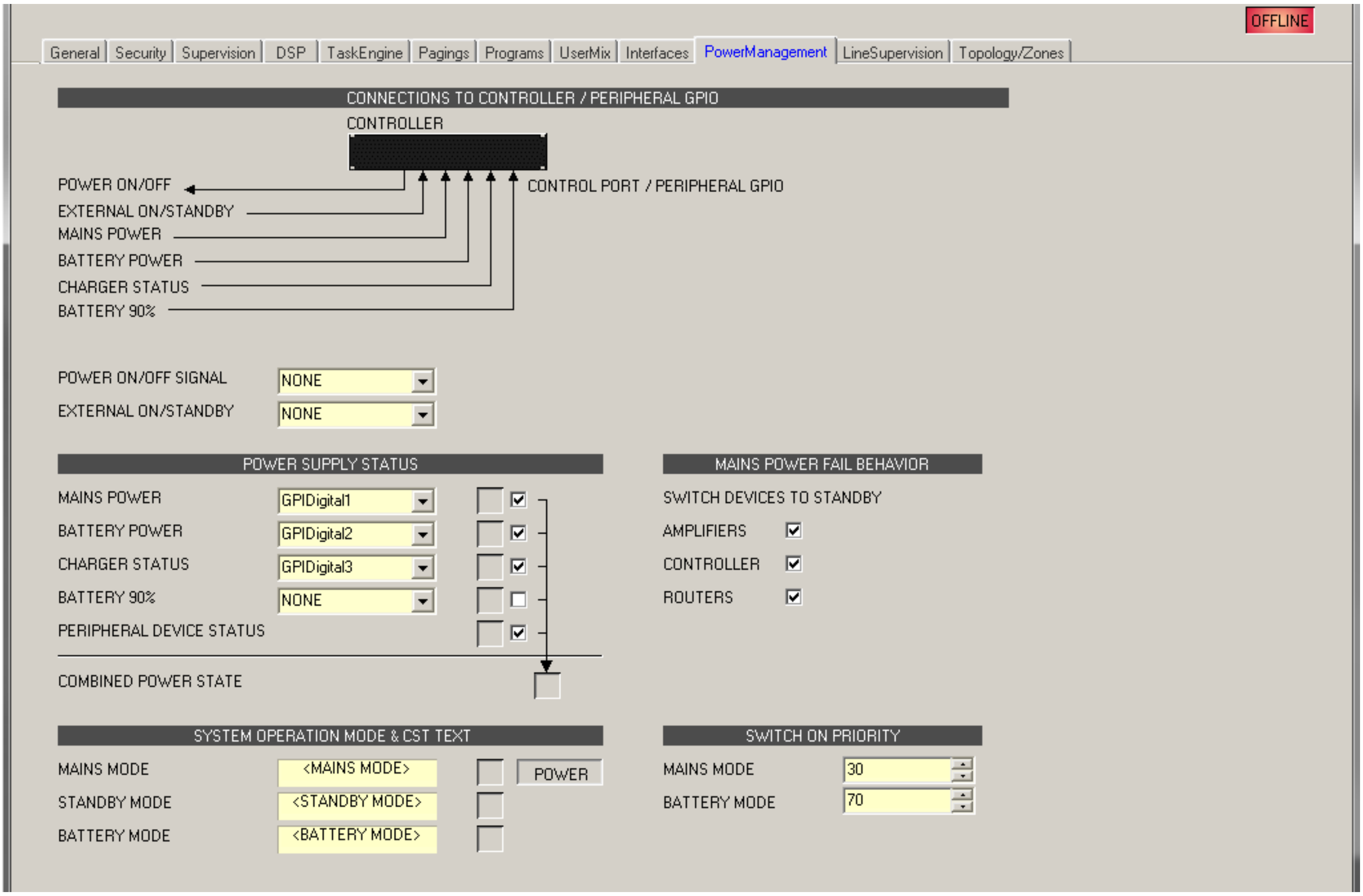


Element	Description
CAN INTERFACE	
BAUD RATE	Transmission rate of the CAN-Bus. All devices on the CAN-Bus must be set to one common transmission rate. **HINT: Editing the CAN BAUD RATE setting is possible in offline mode only.**
CAN STATE	Displays the current CAN-Bus status. Possible indications are: BUS OK, Bus Heavy, Bus Off.
CAN DEVICE LIST	Lists the connected devices.
OPEN INTERFACE	
ENABLE	Set the checkbox to activate the ASCII control protocol of the device.
TCP Port	TCP port of the ASCII control protocol. The default port is 6273.
NUMBER OF LOGIC VALUES	Enter the number of logic values of the task engine to be available via the ASCII control protocol.

NUMBER OF ANALOG VALUES	Enter the number of analog values of the task engine to be available via the ASCII control protocol.
IS ALIVE PERIOD (s)	Enter the is alive period of the ASCII control protocol in seconds.
PASSWORD	If password protection of the ASCII control protocol is required, enter the password here. Repeat the password in the CONFIRM PASSWORD field. Go online (write) to set the password in the device. **HINT: Editing the password setting is possible in offline mode only.**
DPM 4000 INTERFACE	
ENABLE	Set the checkbox to activate the RS-232 interface between a DPM 4000 and the PMX-4CR12.
AUDIO OUTPUT	Indicates the audio output of the PMX-4CR12, that outputs the audio signal to the DPM 4000.
AUDIO INPUT	Select the audio input number of the DPM 4000.
MEMOFLAG BROKEN LINE	The DPM 4000 memo flag selected here is used for supervising the RS-232 connection.
CONTROL IN	
STATE	Displays the control inputs’ current state.
VOLTAGE	Displays the control inputs’ current voltage.
FAULT MON	Set the checkbox of supervised control inputs to activate the supervision.
ACTIVE	Set the upper and lower bound (voltage) of the ACTIVE state of the supervised control inputs.
OK	Set the upper and lower bound (voltage) of the OK state of the supervised control inputs.
CONTROL OUT	
STATE	It is possible to manually change the condition of the control outputs (normally open contact / normally closed contact).

PowerManagement Dialog

The Power Management dialog allows configuring the standby mode of the device in detail.



Element	**Description**
POWER ON/OFF SIGNAL	Select the GPO contact or the virtual TE value for signaling the controllers' operating mode. In standby mode the GPO is open.
EXTERNAL ON/STANDBY	Select the digital GPI or virtual TE value to be used for switching to standby mode.
POWER SUPPLY STATUS	
MAINS POWER	Select the digital GPI or virtual TE value that is used for signaling „mains power OK“. Set the checkbox to monitor this status.
BATTERY POWER	Select the digital GPI or virtual TE value that is used for signaling „battery power OK“.Set the checkbox to monitor this status.
CHARGER STATUS	Select the digital GPI or virtual TE value that is used for signaling „charger status OK“.Set the checkbox to monitor this status.
BATTERY 90%	Select the digital GPI or virtual TE value that is used for signaling „battery status at least 90%“. Set the checkbox to monitor this status.
PERIPHERAL DEVICE STATUS	Set the checkbox to monitor this status.
COMBINED POWER STATE	This LED is green, if all selected power supply status are OK.
SYSTEM OPERATION MODE & CST TEXT	

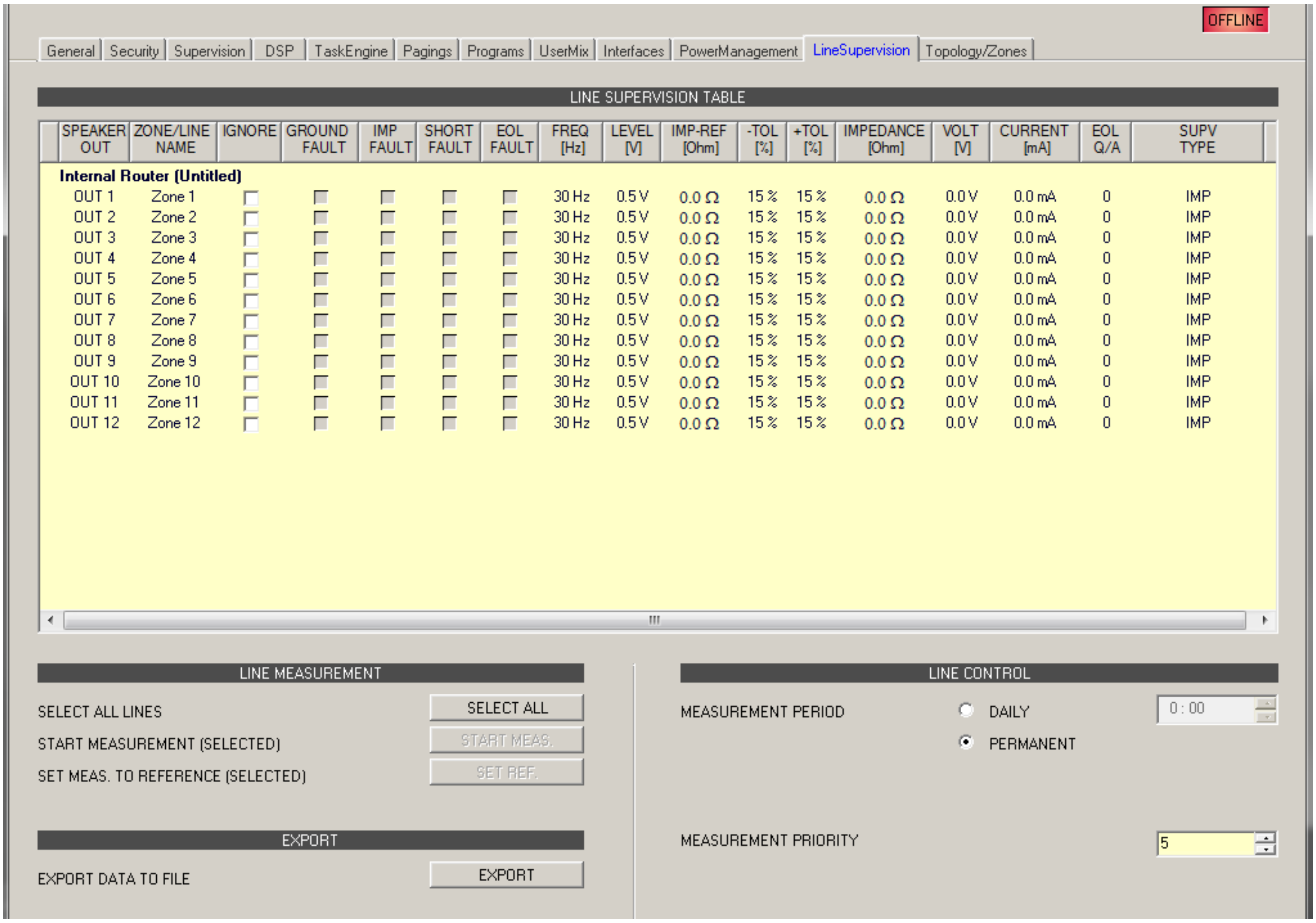
MAINS MODE	If mains power is used for running the system, the Controller is in MAINS MODE and the LED lights green. You can edit the name of this mode in the text field. Press the POWER button to power on/off the device.
STANDBY MODE	If the system is in STANDBY MODE, this LED lights green. You can edit the name of this mode in the text field.
BATTERY MODE	If battery power is used for running the system, the Controller is in BATTERY MODE and the LED lights green. You can edit the name of this mode in the text field.
MAINS POWER FAIL BEHAVIOR	
AMPLIFIERS	Select this option if Amplifiers should switch to standby mode if mains power fails.
CONTROLLER	Select this option if the Controller should switch to standby mode if mains power fails.
ROUTERS	Select this option if Routers should switch to standby mode if mains power fails.
SWITCH ON PRIORITY	
MAINS MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is available.
BATTERY MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is not available (battery mode).

Notice!
The “Power Calculator” tool can be used to calculate the power consumption of the system. The tool can be found in directory “/Tools” or can be requested from the IRIS-Net support team.

Notice!
The properties “Operating Mode” and “Standby LED” can be used for advanced power management configuration via the Task Engine, please refer to section *Properties, page 777*.

LineSupervision Dialog

The Line Supervision dialog allows configuring and control of the Controller line supervision. Line supervision can be done via line impedance measurement method or the End of Line method using EOL 8001 or Plena EOL modules. Please refer to the Controller manual for details about the different measurement methods.



Element	Description
SPEAKER OUT	System internal description of the zone or line.
ZONE/LINE NAME	Description of the zone or line.
IGNORE	Check this checkbox, if the result of the line measurement should be ignored. A error of this zone or line will not be indicated in the system. Regular measurements are carried out anyway. **HINT: If the checkbox is checked a short cut will not be indicated. If the zone is connected via a line relay, the relay will be deactivated.**
GROUND FAULT	This LED lights red, if a ground fault error has occurred.
IMP FAULT	This LED lights red, if the measured impedance is out of the tolerance range.
SHORT FAULT	This LED lights red, if there is a short cut at the zone or line (measured impedance value below 25% of reference value). In this case the system will not start calls or alarms in this zone or line. **HINT: If the zone is connected via a line relay, the relay will be deactivated when there is a short cut (short cut protection for other lines on the same amplifier).**
EOL FAULT	This LED lights red, if a EOL error has occurred.

FREQ [Hz]	Enter the frequency of the measurement signal.
LEVEL [V]	Enter the level of the measurement signal.
IMP-REF [Ohm]	Indicates the impedance reference value of the zone or line.
-TOL [%]	Maximum negative deviation of the impedance value of the zone or line from the reference value, given in percent.
+TOL [%]	Maximum positive deviation of the impedance value of the zone or line from the reference value, given in percent.
IMPEDANCE [Ohm]	Indicates the impedance value of the zone or line of the last successful measurement.
VOLT [V]	Indicates the voltage of the measurement signal of the last successful measurement.
CURRENT [mA]	Indicates the current of the measurement signal of the last successful measurement.
EOL Q/A	Indicates the quantity and addresses of the EOL modules in the zone or line.
SUPV TYPE	Select the supervision method used for the zone. Possible methods are: – IMP = impedance method – EOL = EOL method using EOL 8001 modules – PEOL = EOL method using Plena EOL modules
SELECT ALL	All zone or lines are selected.
START MEASUREMENT (SELECTED).	Starts the line measurement in all selected zones or line.
SET MEAS. TO REFERENCE (SELECTED).	Press this button to store the values of the last measurement as new reference values for the selected zones or lines.
EXPORT DATA TO FILE	All measurement data of the LINE SUPERVISION TABLE are exported to a csv file. Open the file in a spreadsheet for further processing.
DAILY	Set this checkbox, if a daily measurement should be done automatically. Enter the time the measurement should start.
PERMANENT	Set this checkbox, if line measurement should be done permanently.
PRIORITY	Priority of the line measurement signal.

The Line Supervision table is automatically generated from the available zones filled with default values.

Notice!
Use copy & paste to copy configurations from one element to another element in the line supervision table.

IMPEDANCE METHOD

The values of frequency, level and tolerance can be edited and adapted to the real conditions. To generate the reference values a first line measurement must be performed, the resulting measurement values are stored as reference values. The measurement of the lines and the comparison with the reference values is done automatically either permanently or every day at the scheduled time if the line is not busy. Each audio signal on the line interrupts the line measurement. The measurements will be continued automatically if the line is free again.

Notice!

Impedance reference values (IMP-REF) shall be measured and set for all used loudspeaker lines. The reference values are necessary not only if supervision type IMP is selected, but also for short circuit detection if supervision type EOL or PEOL is selected. The reference values are further needed for an impedance measurements triggered by an amplifier overload detection.

EOL METHOD

To enable the EOL supervision for a zone or line in the column EOL Q / A in the first line the number of EOL modules connected to the line must be entered, in the following lines the addresses of the modules must be entered. Enter 0 to disables the EOL method for the corresponding line.

Notice!

For power supply of the EOL modules a pilot tone is required, so the pilot tone generator of the power amplifier shall be activated.

Topology/Zones Dialog

The Topology/Zones dialog window allows configuration of Topologies and Zones. Zones are configured in a topology, each zone can be selected to be member of a Group.

IRIS-Net offers basic validation checks if the connections are valid. If a rule is not followed correctly, the connection line turns red.

The following rules apply:

- One amplifier output can be connected to one or more router cluster inputs in parallel please refer to Line topology. Only identical inputs of router clusters can be connected in parallel, e.g. if a amplifier output is connected with the AMP IN1 input of one router cluster, the amplifier output can only be connected to AMP IN1 inputs of other router clusters. The same is valid for the router cluster input AMP IN 2. So it is not allowable to connect different types of router cluster inputs to one amplifier output.
- One amplifier output can always be used for one topology only. E.g. if a amplifier output is connected to a router cluster set to 1-in-N topology, the amplifier output can not be connected to a router cluster using another type of topology (e.g. 2-in-N).
- If the two output channels of an amplifier are used for a 2-in-N or Program/Call topology and the outputs are connected to more than one router cluster, the outputs have to be connected to the same input (e.g. always AMP IN 1) of the router clusters.
- If a 2-in-N or Program/Call topology is selected, going online is not possible if the identical amplifier output is connected to both inputs.
- Do not connect an amplifier output to a “regular” input of a router cluster (AMP IN 1, AMP IN 2) and a spare amplifier input (S1, S2,) at the same time.
- If router clusters with EOL8001 supervision are used, always connect one amplifier output to router clusters of one device only.

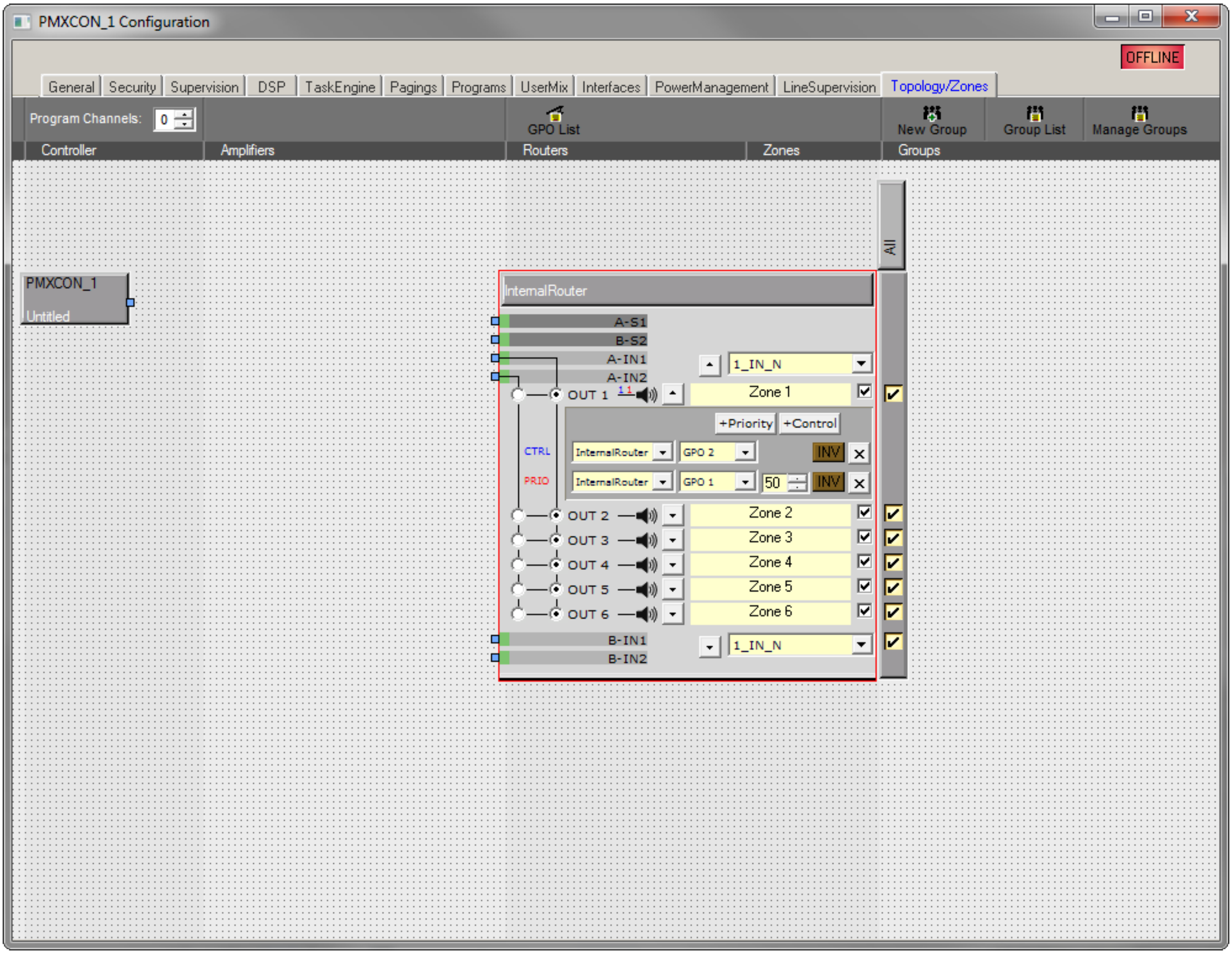
Notice!
If router clusters with EOL8001 supervision are used, switching router clusters within one device is possible. This allows creating 1-in-24 or 2-in-24 topologies with EOL 8001 supervision.

- If a 1-in-N topology is selected, the connections of the router cluster inputs (IN1 or IN2) have to be set correctly. E.g. every loudspeaker output has to be connected to a router cluster input, and the router cluster input has to be connected to a amplifier output.

For spare amplifier switching, the following rules apply:

- Automatic spare amplifier switching can be activated for every amplifier channel in the system. One spare amplifier channel is possible for an amplifier channel connected to the AMP IN 1 or AMP IN 2 input of a router cluster. The spare amplifier channel as to be connected to the S1 or S2 input of the same router cluster.
- If an amplifier output channel is connected to the inputs of more than one router clusters in parallel and should be backed up by a spare amplifier, the channel of the spare amplifier has to be connected to the same router clusters in parallel.

If a 2-in-N or Program/Call topology is selected and the amplifier should be backed up by a spare amplifier, two spare amplifier channels are required. It is not allowed to backup both channels of the topology with one spare amplifier channel only.



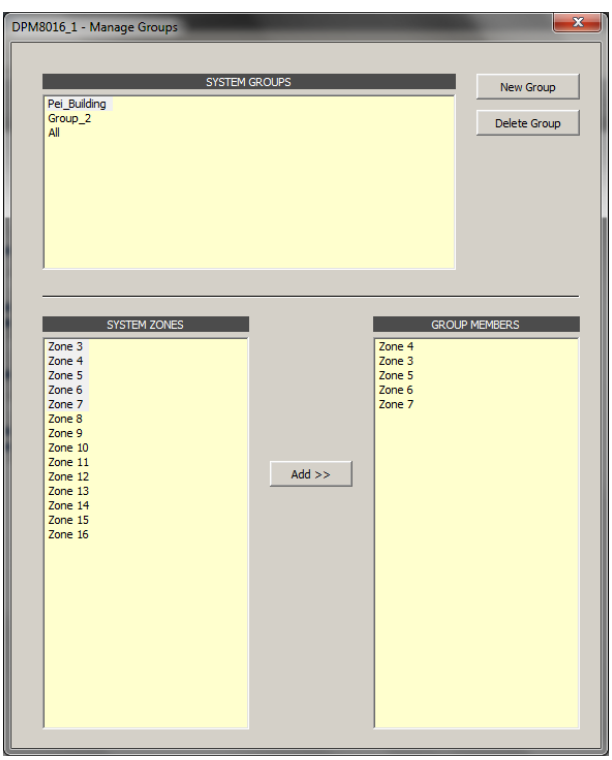
Icon bar

Element	Description
Program Channels	Select the number of program channels in the Controller.
GPO List	Click on this button to generate a CSV report of all GPOs configured in the system.
New Group	Click on this button to create a new Group. The “All” group, including all zones, is created automatically. For every new group the zones can be selected via the checkboxes in the group column.
Group List	Click on this button to generate a CSV report of all Groups configured in the system. The report includes the caption and object id of systems zones and the assignment of zones to system groups.
Manage Groups	Click on this button to open the Manage Group Dialog. This dialog allows to add or delete Groups and to add or remove Zones from a selected Group.

Element	**Description**
1_IN_N, 2_IN_N, PROG_CALL	Select the topology for the 2-in-6 cluster.
	Press this button to minimize or maximize the zones or relays dialog.
Zone 1	Enter a name for the zone.
+Priority	Press this button to add a priority relays to the zone. Note: Up to 2 priority relays can be configured in a zone.
+Control	Press this button to add a control relays to the zone. Note: Up to 2 control relays can be configured in a zone.
Device dropdown	Select the device to be used for controlling the control or priority relay.
GPO dropdown	Select the GPO (of the selected device) for controlling the control or priority relay.
 50	This control allows setting the priority value of a priority relay.
INV	Press the INV button to invert the status of the control or priority relay.
X	Press this button to delete the corresponding priority or control relay.

Manage group dialog

This dialog allows to create, edit or delete groups. It is also possible to add or remove zones from a selected Group. To remove a zone from a group, select the zone in the GROUP MEMBERS section and press the delete button.



Element	**Description**
SYSTEM GROUPS	Lists all groups of the system.
New Group	Press this button to create a new group.

Element	Description
Delete Group	Press this button to delete the group selected in the SYSTEM GROUPS list.
SYSTEM ZONES	Lists all zones of the system.
Add >>	Adds the zones selected in the SYSTEM ZONES list to the group selected in the SYSTEM GROUPS list.
GROUP MEMBERS	Lists the zones currently included in the group selected in the SYSTEM GROUPS list.8

Properties

OPERATION MODE

The “PMXCON_1.System.PowerManagement.OperatingMode” property allows setting the current operation mode of the PMX-4CR12 and connected devices. High priority signals prevent changing into standby mode. Following settings are available:

Value	Description
0	Switch PMX-4CR12 in standby mode
1	Switch PMX-4CR12 in operating mode

HINT: The mode of peripheral devices connected to the PMX-4CR12 is set automatically.

STANDBYLED

The Standby LED of the PMX-4CR12 lights, when the device is in standby mode. The corresponding property “PMXCON_1.System.Info.StandbyLED” can be used to query the current mode.

Value	Description
0	PMX-4CR12 is in operating mode
1	PMX-4CR12 is in standby mode

PMX-4R24

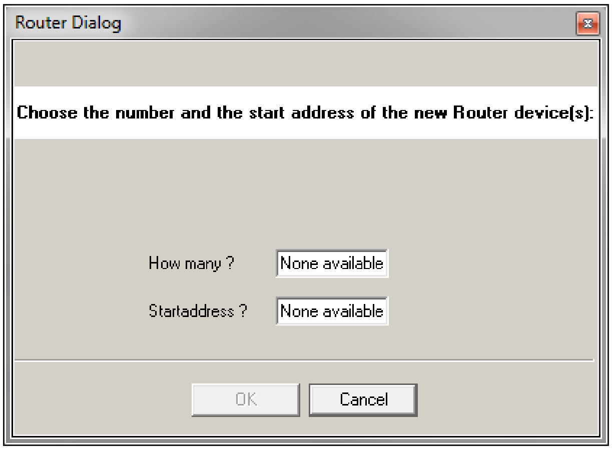
The PMX-4R24 24 Zone Router is a zone extension for the PROMATRIX 6000 system. The PMX-4R24 adds 24 zones, 20 GPIs, 24 GPOs and 2 control relays to the system and is controlled and supervised via the CAN bus by the PMX-4CR12 (Controller). Up to 20 external routers can be connected to one controller. One router can handle up to 4000 W speaker load. The maximum load of one zone is 500 W.

The zone indicator lights on the front indicate the current status of every zone:

- Green: Zone in use for non emergency purpose
- Red: Zone in use for emergency purpose
- Yellow: Zone fault detected
- Off: Zone in idle condition

PMX-4R24 Device

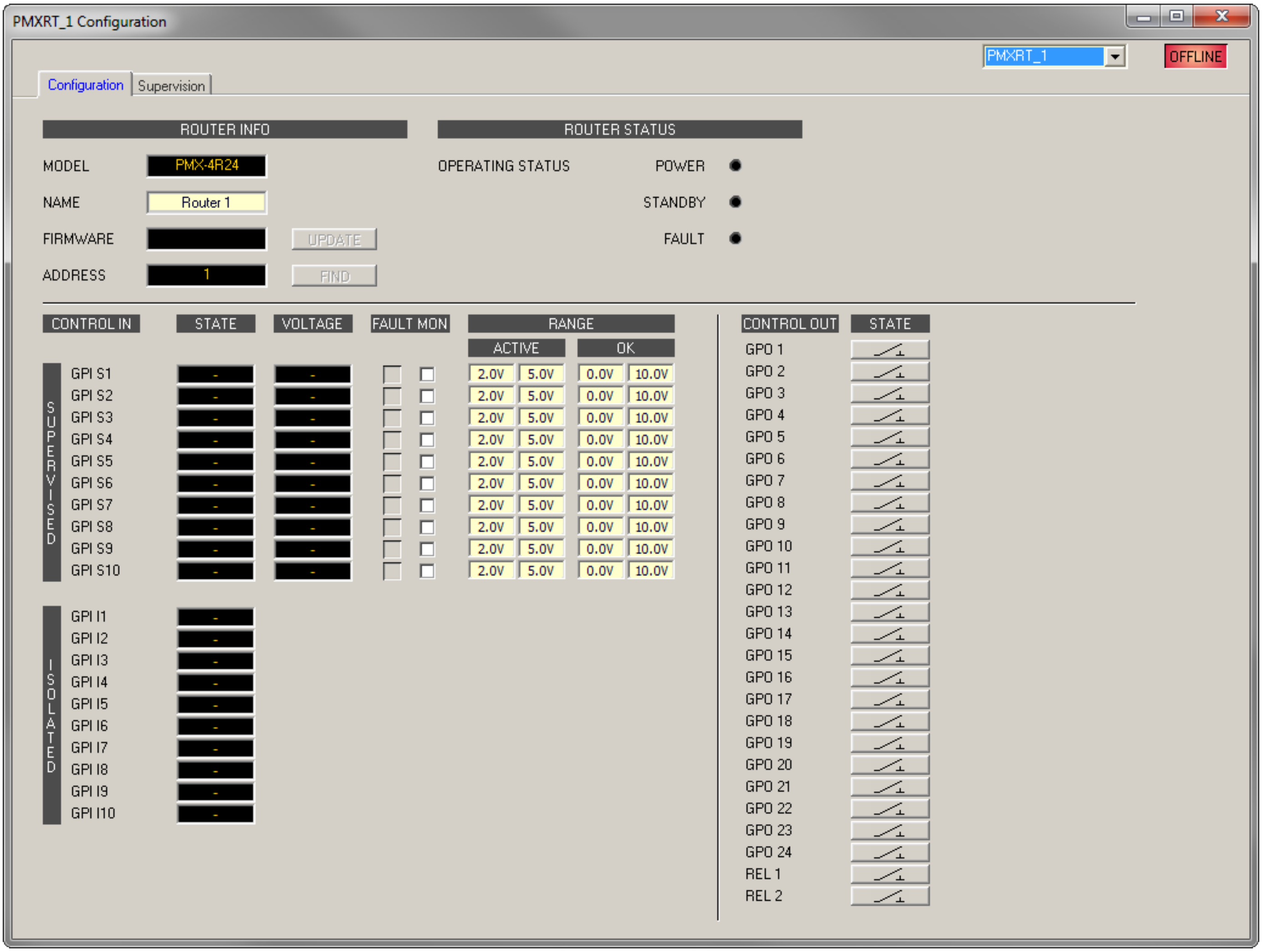
Start by creating a PMX-4R24 Device in your IRIS-Net project. Drag a PMX-4R24 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.

Configuration Dialog

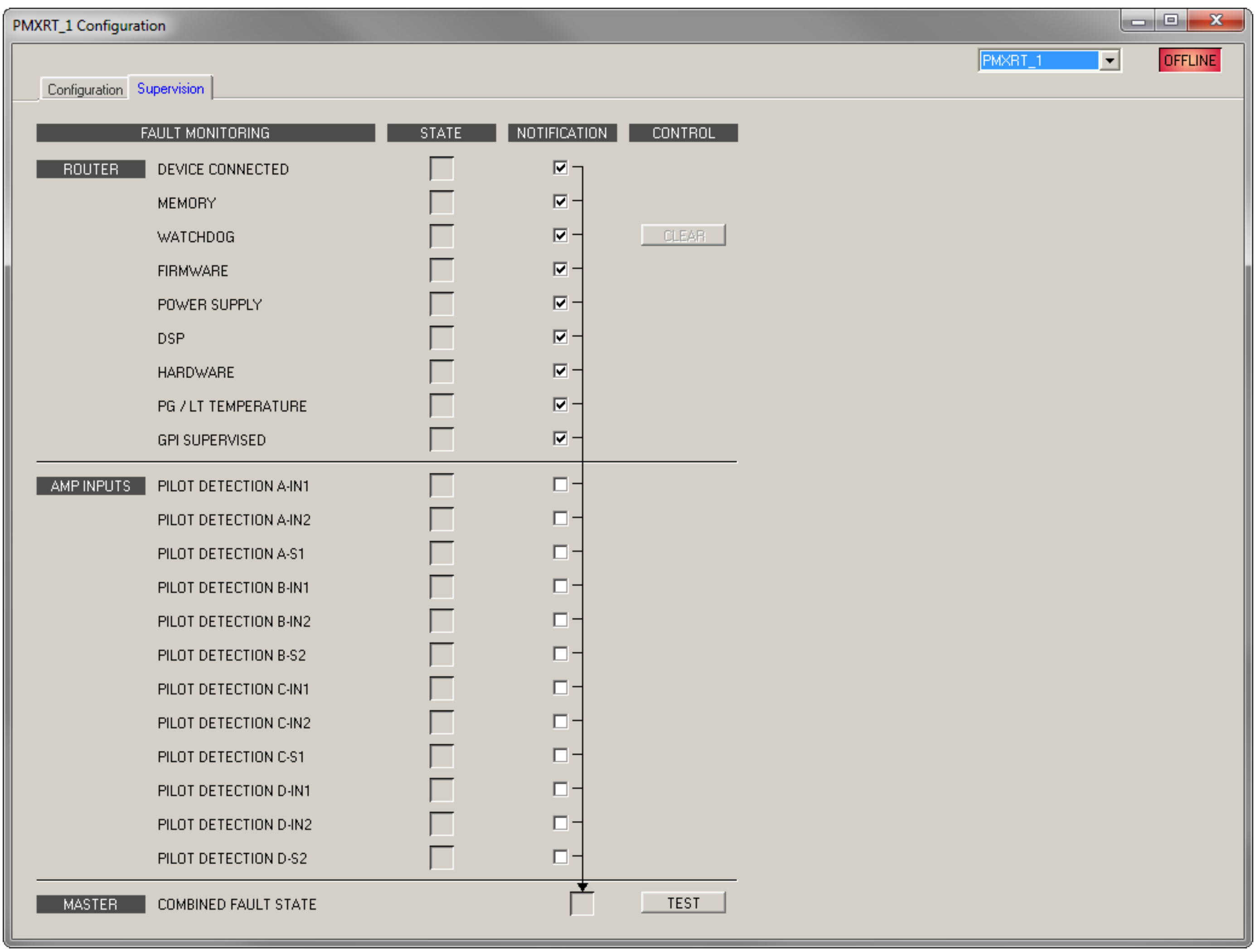


Element	Description
MODEL	Indicates PMX-4R24, so you know the model of the device.
NAME	IRIS-Net internal device name of the Router.
FIRMWARE	Indicates the firmware version of the Router when on-line.
	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
ADDRESS	Indicates the CAN address of the device.
FIND	Press the find button to activate the find function of the device.
OPERATING STATUS	Indicates the operating status of the Router.
CONTROL IN	
STATE	Displays the control inputs’ current state.
VOLTAGE	Displays the control inputs’ current voltage.

FAULT MON	Set the checkbox of supervised control inputs to activate the supervision.
ACTIVE	Set the upper and lower bound (voltage) of the ACTIVE state of the supervised control inputs.
OK	Set the upper and lower bound (voltage) of the OK state of the supervised control inputs.
CONTROL OUT	
STATE	It is possible to manually change the condition of the control outputs (normally open contact / normally closed contact).

Supervision Dialog

The Supervision window shows the condition of the PMX-4CR12. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT indicator light on the front panel of the controller lights, the FAULT relay opens and a signal sound.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the corresponding HOLD checkbox is unchecked.
LOG	
CONTROLS	

ROUTER

DEVICE CONNECTED	CAN connection between Controller and Router broken.
MEMORY	Memory error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
POWER SUPPLY	Error in the power supply of the device.
DSP	Error in the digital signal processing (DSP) of the device.
HARDWARE	Hardware error.
PG / LT TEMPERATURE	Temperature overload of the device.
GPI SUPERVISED	Voltage at supervised GPI out of range.

AMP INPUTS

PILOT DETECTION x-IN1	Missing pilot tone at input 1 of cluster A or B.
PILOT DETECTION x-IN2	Missing pilot tone at input 2 of cluster A or B.
PILOT DETECTION A-S1	Missing pilot tone at spare input 1 of cluster A.
PILOT DETECTION B-S2	Missing pilot tone at spare input 2 of cluster B.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

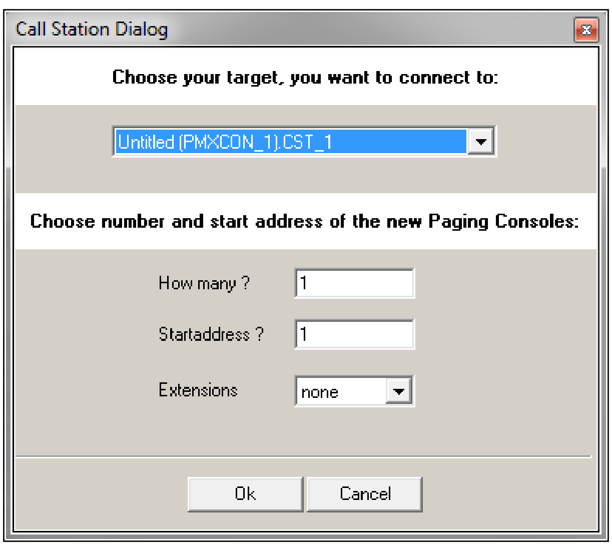
PMX-15CST

The PMX-15CST is a call station for the PROMATRIX 6000 system. As standard, the call station has a gooseneck microphone with pop shield and permanent monitoring, a total of 20 buttons, an illuminated LC display, and an integrated loudspeaker. The call station can be modified to suit the user's requirements by connecting up to five PMX-20CSE call station extensions, each with 20 customizable selection buttons.
Other properties:

- Five menu/function keys (pre-programmed) – one green or one yellow indicator light per button
- 15 selection buttons (customizable) – two indicator lights (green/red) per button
- Label with transparent covering – the label can be changed at any time
- Can be used as a standing or desk/rack flush-mounted device
- Internal monitoring with error logging – complying with all relevant national and international standards
- Easy configuration – use of the Configuration Wizard or IRIS-Net software

PMX-15CST Device

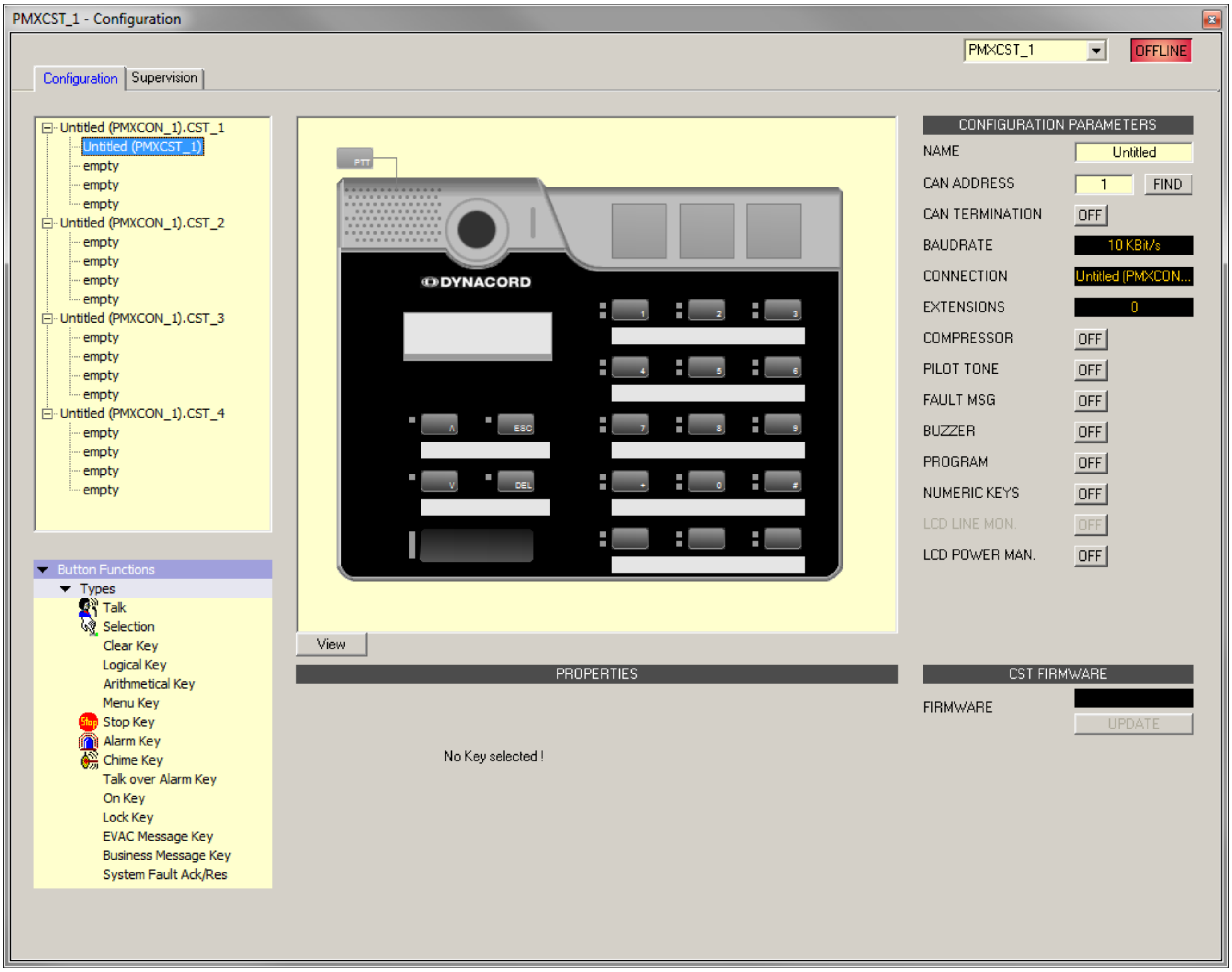
Start by creating an PMX-15CST device in your IRIS-Net project. Drag an PMX-15CST from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



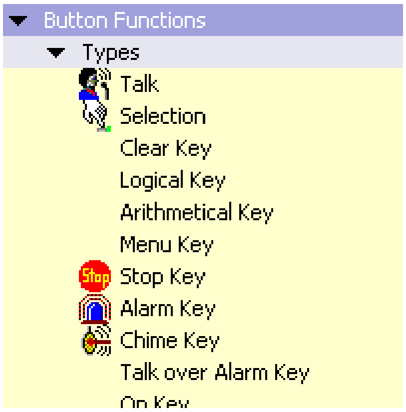
Select the call station bus the device is connected to.
Specify the desired number of devices, the address of the call station and number of call station extensions (it is not possible to add extensions to a call station kit). Click on the OK button to accept these settings.
The specified number of Call Stations will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on a Call Station device icon opens the configuration dialog window. Double clicking on a device for the first time will open the Configuration dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the Call Station device icon.

Configuration Dialog

This page allows making basic settings and retrieve information, for example of button functions, network settings, device name, firmware version, etc.



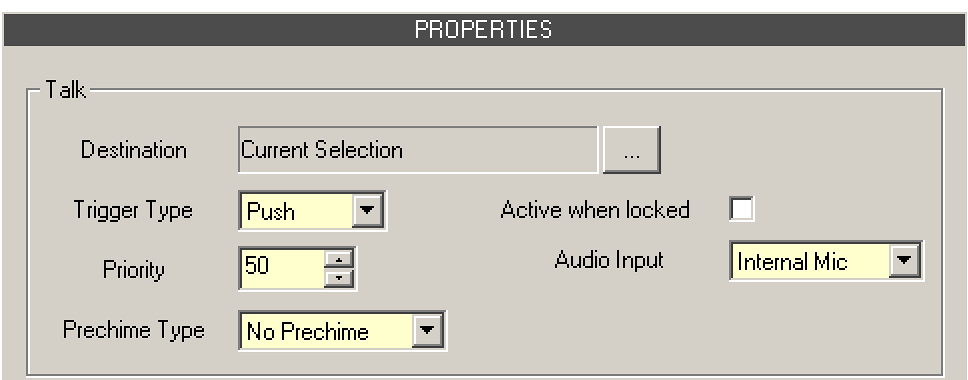
Element	Description
	When there are several call stations connected to the CST busses of the Controller you can select the call station (kit) to configure here.

	Select the desired button type and drag it from this dialog box onto the button of a call station or a call station extension. Detailed information about different types of buttons is provided on the following pages.
ONLINE OFFLINE	The Online / Offline indicator signals whether the call station is included in the network or off-line. The red OFFLINE indicator signals that the corresponding call station is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding call station is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
NAME	Name of the device.
CAN ADDRESS	Displays and lets the user enter the CAN address of the call station. Left-click in the field and enter the desired address in the range from 1 to 16. The entered value is adopted by pressing RETURN. The entered address has to match the setting in the call station's menu and may only exist once. When adding new call stations to an IRIS-Net project, CAN addresses are automatically assigned in ascending order.
FIND	When pressing this button, the backlight of the call station's LCD screen blinks regularly in quick succession. The status indicator of the call station Device in IRIS-Net blinks at the same time. This function serves for checking communication and for identification or search of a call station in a larger system.
CAN TERMINATION	Press this button (ON) to activate the internal termination resistor of the CAN bus in the call station.
BAUDRATE	The baud rate of the call station.
CONNECTION	Name of the Controller the call station is connected to.
EXTENSION	Number of Call Station extensions.
COMPRESSOR	Press this button (ON) to activate the internal compressor of the call station.
PILOT TONE	Press this button (ON) to activate the pilot tone supervision of the call station. **HINT: When using the pilot tone supervision only one call station can be connected to a CST bus.**
FAULT MSG	Press this button (ON) if error messages should be indicated in the LC-display of the call station.

BUZZER	Press this button (ON) if errors should be signaled via the integrated buzzer.
PROGRAM	Press this button (ON) if the Program Assignment menu should be accessible in the LC-display of the call station.
NUMERIC KEYS	Press this button (ON) to allow numeric entry of zone numbers.
LCD POWER MAN.	Press this button (ON) to indicate power management states in the display of the call station.
View	Switching between the following views of a call station and (if existing) call station extensions: – Scroll View – Overall View – Selective View
FIRMWARE	Indicates the firmware version of the Call Station when on-line.
UPDATE	Press this button to update the firmware of the call station. **NOTE: The default password for the firmware update is “0000”.**

Talk

A switch of the type "Talk" allows configuring a TALK button. Specific Zones and/or Groups can be pre-selected for this key. Pressing the button on the call station automatically selects the Zones and/or Groups in which the spoken message is being heard.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Trigger (triggers a function)
Priority	Select the button's priority (0 to 9).
Audio Input	Select one of the following audio sources for the announcement: – Internal Mic – External Mic – External Line

Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Prechime Type	Select the desired type of pre-gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – No Prechime – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime

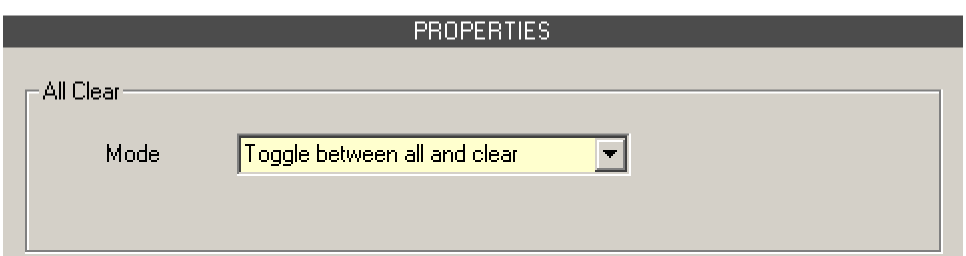
Selection

A switch of the type "Selection" allows configuring a SELECT button. Pressing the button on the call station selects the Zones and/or Groups that have been configured here.

Element	Description
Destinatio n	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.

Clear Key

A switch of the type "Clear Key" allows configuring an ALL/CLEAR button. Pressing the button on the call station selects or deselects all Zones and/or Groups.



Element	Description
Mode	Select the function that is to be executed when pressing the button on the call station: – Toggle between all and clear = Each press of the button alternately selects or deselects all Zones and/or Groups. – Select All = Pressing the button selects all Zones and/or Groups of the whole system. – Deselect All = Pressing the button deselects all Zones and/or Groups.

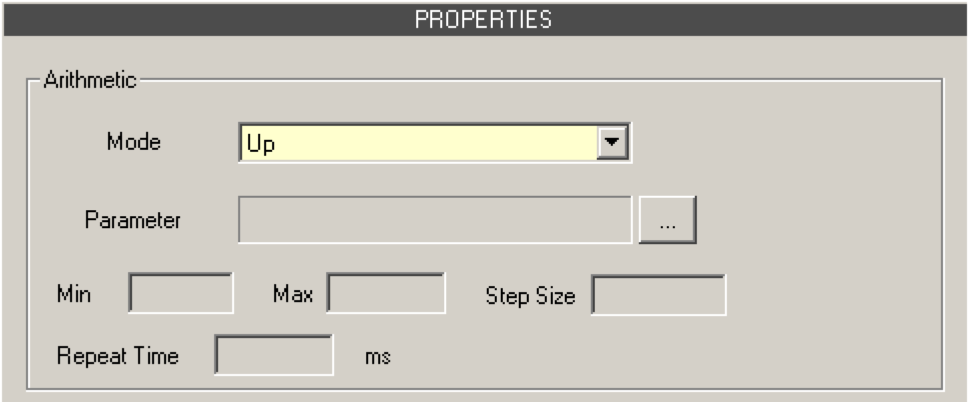
Logical Key

A switch of the type "Logical Key" allows setting the value of a logic variable (0 or 1). Pressing the button on the call station sets the logic variable to the desired value. The adjacent LED is operated according to the resulting parameter.

Element	**Description**
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Set Value = sets the value of the logic variable to "1". It remains "1", even after the button is being released. – Reset Value = sets the value of the logic variable to "0". It remains "0", even after the button is being released. – Push = sets the value of the logic variable to "1", but only as long as the button is being pressed. – Toggle = inverts the value of the logic variable each time the button is being pressed. – LED only = indicates the value of the logic variable, the value is not changed by the button
On	Select the LED of the button that should indicate the value „1“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Off	Select the LED of the button that should indicate the value „0“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Parameter	The logic variable whose value is being changed.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Arithmetical Key

A switch of the type "Arithmetical Key" allows changing the value of a numerical variable. Pressing the button on the call station either increases or decreases the value of the numerical variable.



Element	Description
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Up = increases the value of the numerical variable – Down = decreases the value of the numerical variable
Parameter	The numerical variable whose value is being changed.
Min	The lower limit of the value range. Using the "Down" mode decreases the value of the numerical variable till down to this value.
Max	The upper limit of the value range. Using the "Up" mode increases the value of the numerical variable till up to this value.
Step Size	Lets the user enter the step width by which the value is to be changed when pressing the button on the call station.
Repeat Time	Lets the user enter a value for the time interval in milliseconds after which (when keeping the button pressed) the numerical value is being changed by the set step width art any one time. Entering "0" changes the value only once, even when keeping the button pressed over a longer period of time.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

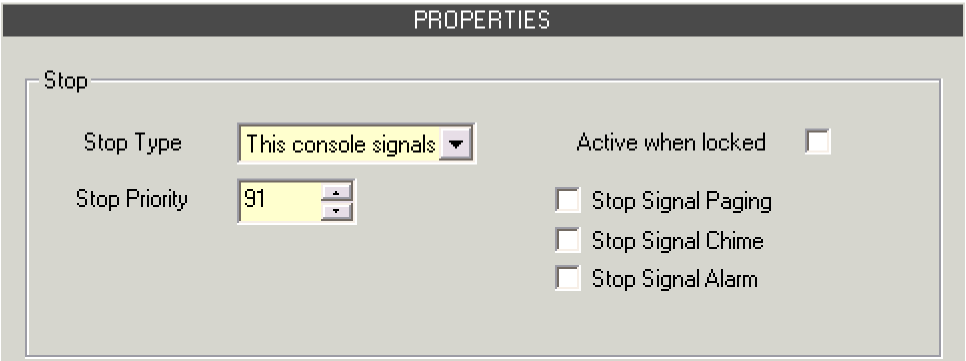
Menu Key

A switch of the type "Menu Key" displays the menu on the LCD screen of a call station.

Element	Description
Jump to	Select the position in the menu structure that is to be displayed

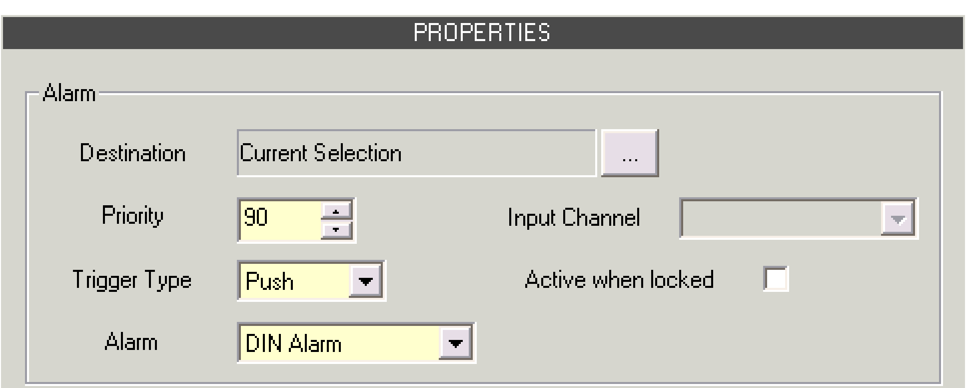
Stop Key

A switch of the type "Stop" allows canceling a process that is currently running on the system.



Element	Description
Stop Type	Select the function that is to be executed when pressing the button on the call station: – This concsole signals (local actions) = stops only the types of actions that have been launched from this specific call station – System signals = stops all selected types of actions system-wide, even if they have not been launched from this specific call station
Stop Priority	Select the maximum priority for the signals that will be stopped when pressing the button on the call station.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Stop Signal Paging	Pressing the button on the call station stops pagings.
Stop Signal Chime	Pressing the button on the call station stops chimes.
Stop Signal Alarm	Pressing the button on the call station stops alarms.
Stop Signal Text	Pressing the button on the call station stops signal texts.

Alarm key
A switch of the type "Alarm" allows starting an Alarm on the system.

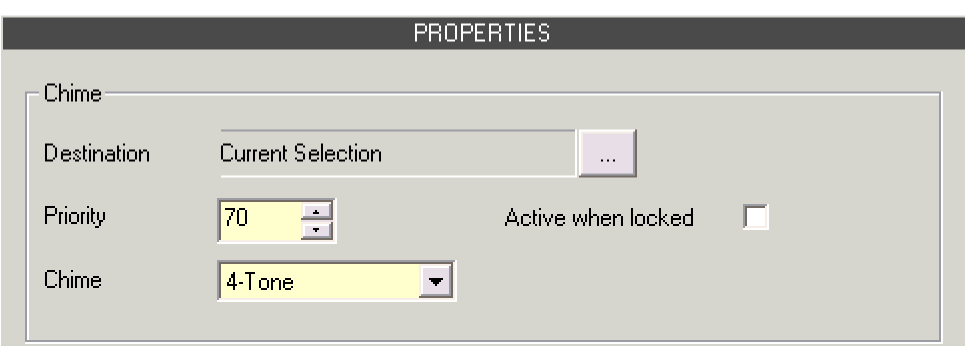


Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.

Priority	Select the alarm priority (0 to 100).
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger (triggers a function)
Alarm	Select the desired signal that is to be used for alarming: – Extern – DIN Alarm – Slow Whoop – Siren – Two-Tone Alarm – Telephone Alarm – Ship Alarm 1 – Ship Alarm 2 – Ship Alarm 3 – Ship Alarm 4 – Ship Alarm 5 – Ship Alarm 6 – Ship Alarm 7 – Ship Alarm 8 – Ship Alarm 9 – Ship Alarm 10 – Ship Alarm 11 – Ship Alarm 13 – Ship Alarm 14 – Ship Alarm 15 – Ship Alarm 16 – Ship Alarm 17
Input Channel	Enter the audio input at which the externally generated alarm signal is present.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Chime Key

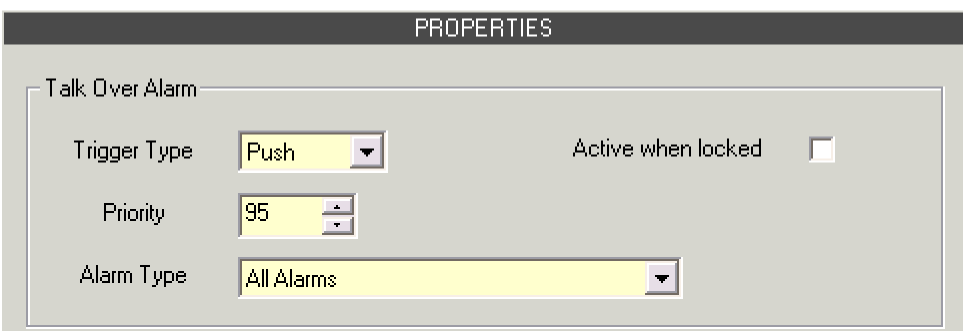
A switch of the type "Chime Key" allows the launch of a gong (chime) signal in the system.



Element	**Description**
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the chime priority (0 to 100).
Chime Type	Select the desired type of gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Talk over Alarm Key

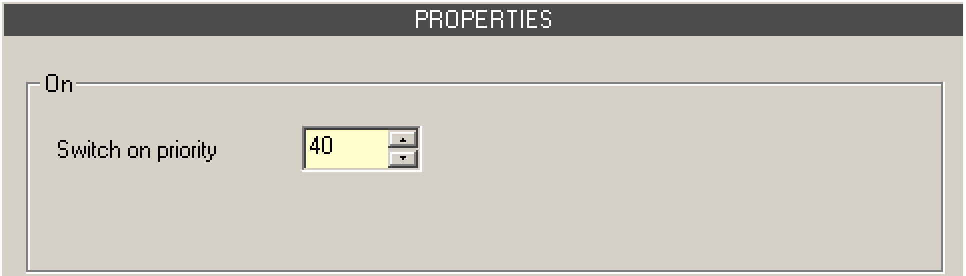
A switch of the type "Talk over Alarm Key" allows making an announcement during an alarm. During the announcement the alarm signal is off, and is started again after the announcement.



Element	**Description**
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states)
Priority	Select the priority (0 to 100) of the announcement. Must be higher than the alarm signal priority.
Alarm Type	Select the alarm type.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

On Key

A switch of the type “On” allows switching the PROMATRIX 8000 system on or off (standby) using a button of the call station.



Element	Description
Switch on priority	Select the priority (0 to 100) of the button.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Lock Key

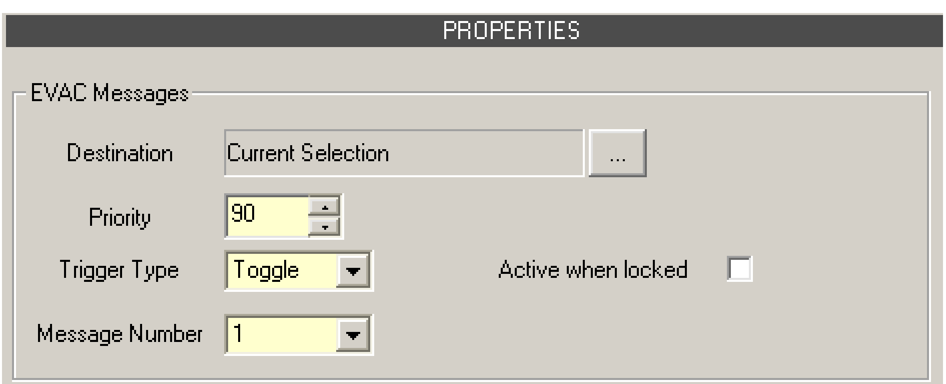
A switch of the type “Lock” allows locking the buttons of the call station. When assigned to a selection button the password set in the Security tab of the Controller has to be entered at the call station..

Notice!

If a button should stay active even if the call station is locked, the „Active when locked“ checkbox of this button has to be selected.

EVAC Message Key or Business Message Key

A switch of the type "EVAC Message Key" or “Business Message Key” allows starting a prerecorded message of type EVAC ore Business Message from the Message Manager.

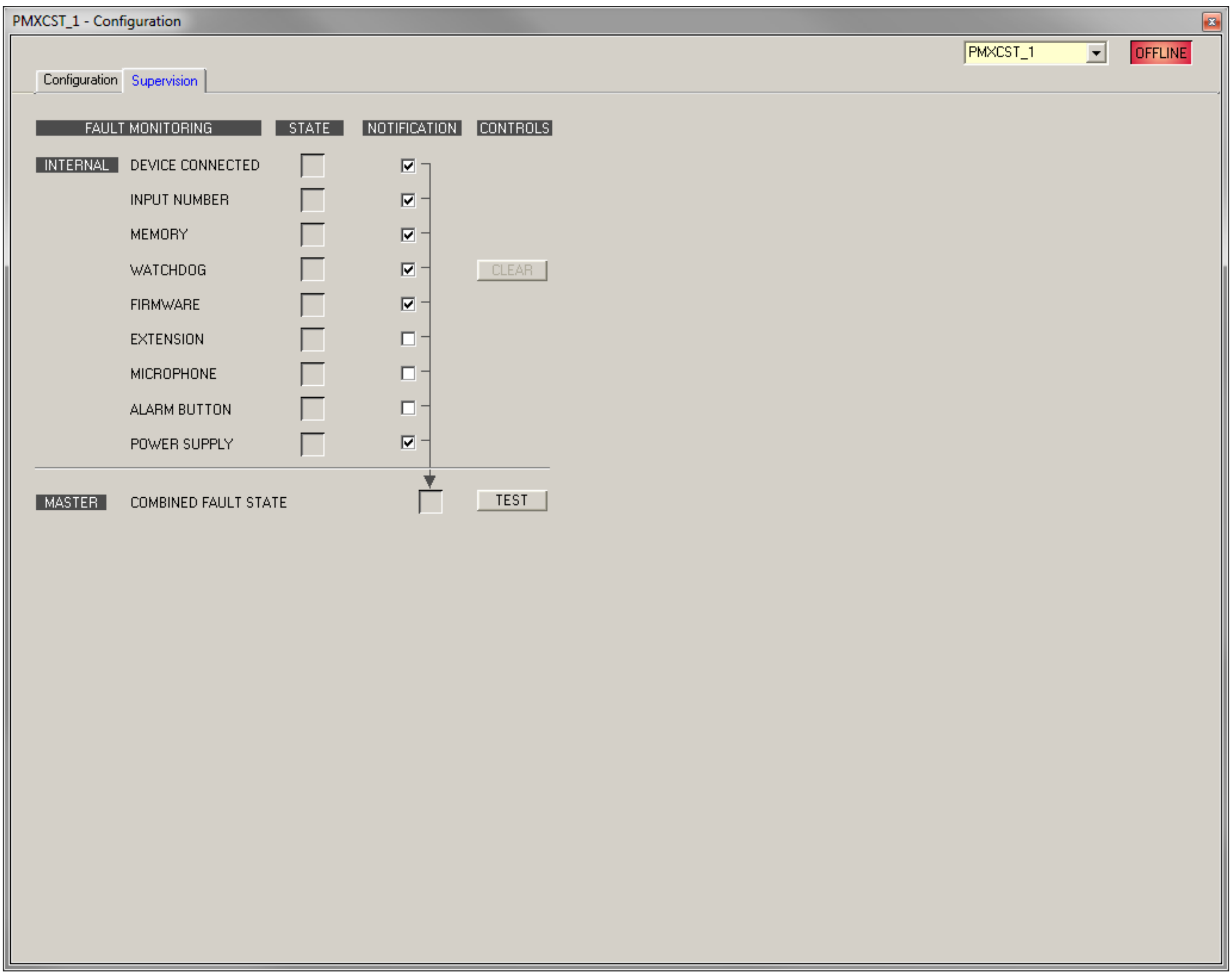


Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the priority (0 to 100) of the message.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger

Message Name	Select the message by name.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Loop	Select this checkbox to automatically repeat the selected message.

System Fault Ack/Res
A switch of the type “System Fault Ack/Res” allows to acknowledge or reset a system fault that is indicated at the call station. This type can be assigned to the DEL button only.

Supervision Dialog



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox NOTIFICATON is ticked, the COMBINED FAULT STATE is set at the same time and the FAULT indicator light at the Call Station lights.
DEVICE CONNECTED	The CST bus connection between Controller and Call Station is broken.
INPUT NUMBER	The call station is not connected to the correct CST bus.
MEMORY	Memory error in the Call Station.
WATCHDOG + CLEAR	Watchdog error in the Call Station. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the Call Station is too old.
EXTENSION	The number of call station extensions is to high or the addresses of the extensions are not correct.
MICROPHONE	Microphone error in the Call Station.
ALARM BUTTON	Supervision fault of the alarm button or the key switch.
POWER SUPPLY	Power supply out of range.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

PMX-CSK

Please refer to section *PMX-CSK Call Station Kit, page 800*.

PMX-2P500

The PMX-2P500 class-D amplifier is a 2 × 500 W professional audio amplifier for evacuation purposes. It can be operated from both the mains and a DC supply. The output voltage is galvanically insulated and is constantly monitored for ground fault. An energy-saving mode and temperature-controlled fans reduce energy consumption and noise levels. The control and monitoring functions are performed via CAN bus. This amplifier is designed for operation in an emergency evacuation system. The amplifiers are usually controlled via a controller and configured using IRIS-Net.
The power amplifier has the following features:

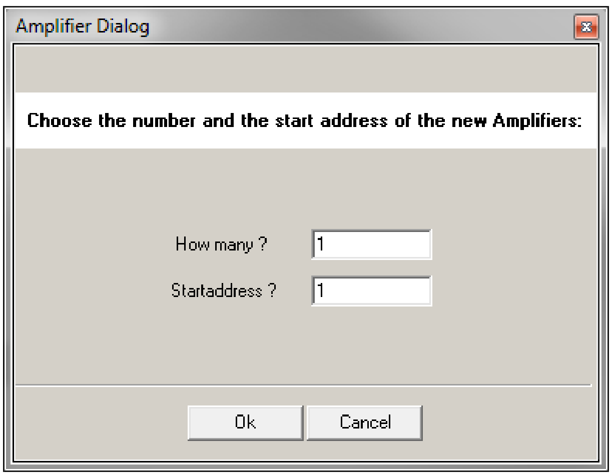
- Floating 100 V or 70 V power outputs
- High efficient amplifier blocks in class-D technology
- Outputs idling and short circuit-protected
- Mains operation 120–240 V (50/60 Hz) and/or 24 V DC emergency backup
- Electronically balanced inputs
- Temperature monitoring function
- Pilot tone and ground fault monitoring function via PMX-4CR12 controller or PMX-4R24 router
- Processor control of all functions
- Monitoring of the processor system via watchdog circuit
- Non-volatile FLASH memory for configuration data
- Internal monitoring function

– Integrated audio relays
– Line monitoring function

The power amplifier is processor-controlled and equipped with extensive monitoring functions. Line monitoring for the CAN bus and for audio transmission allows line interruptions and short-circuits to be detected and indicated to the user.

PMX-2P500 Device

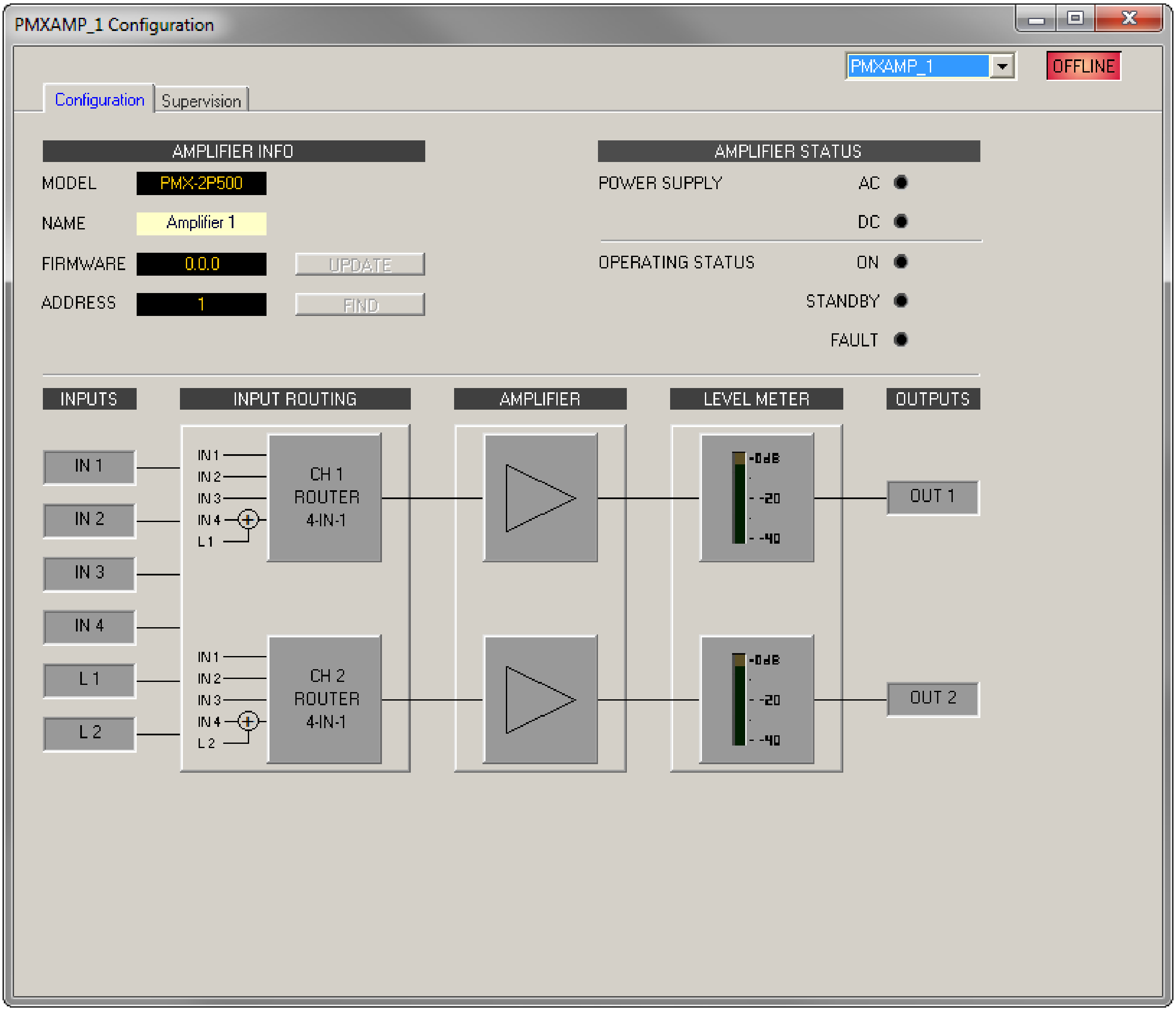
Start by creating an PMX-2P500 device in your IRIS-Net project. Drag an PMX-2P500 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.

Configuration Dialog

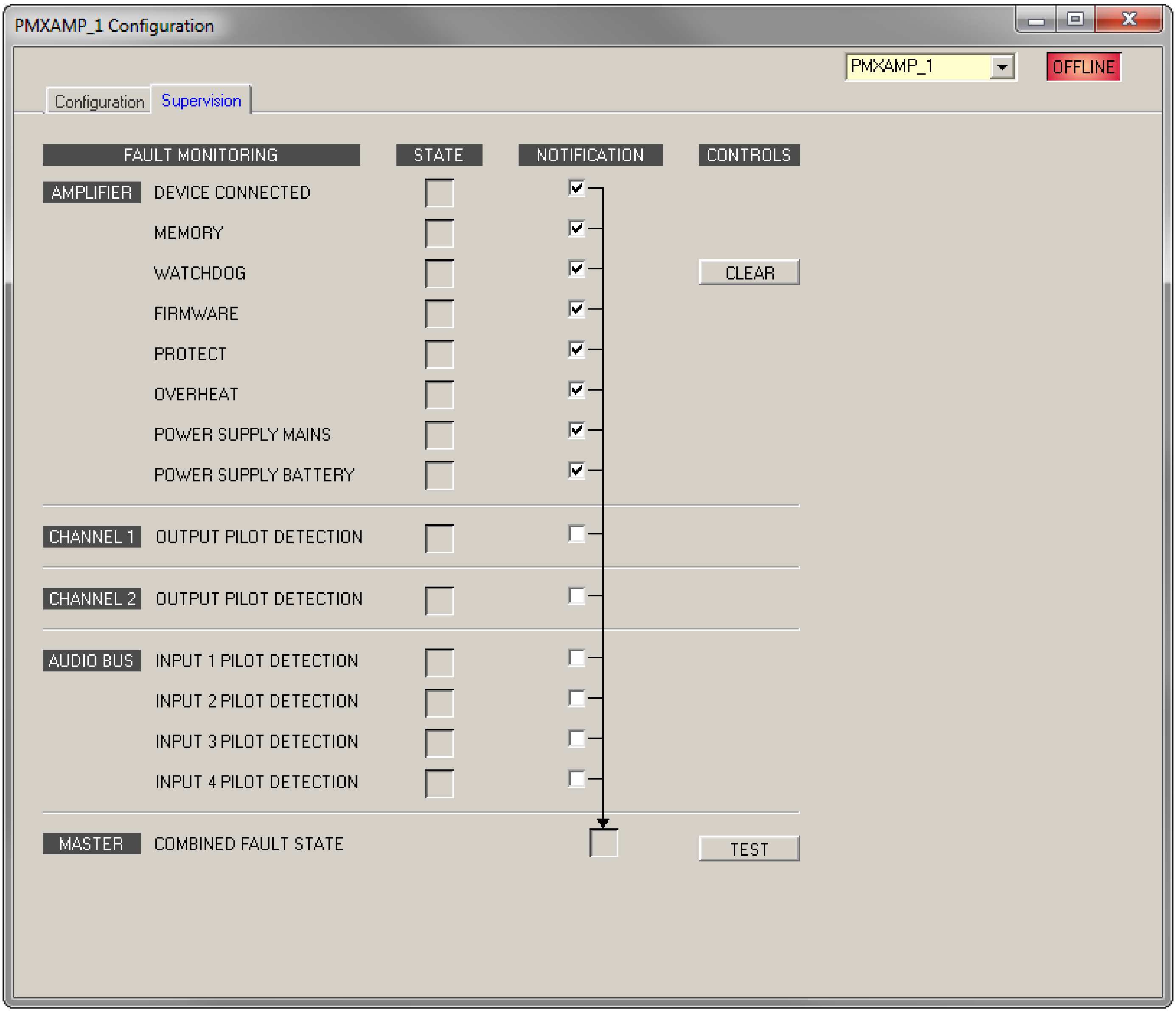


Element	Description
MODEL	Indicates PMX-2P500, so you know the model of the device.
NAME	IRIS-Net internal device name of the Amplifier.
FIRMWARE	Indicates the firmware version of the Amplifier when on-line.
UPDATE	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
ADDRESS	Indicates the CAN address of the device.
FIND	Press the find button to activate the find function of the device.
POWER SUPPLY	Indicates the status of the AC or DC supply voltage.

OPERATING STATUS	Indicates the operating status of the Amplifier.
LEVEL METER	The level meters of the two output channels indicate the signal level of the audio output signal.

Supervision Dialog

The Supervision tab shows the condition of the PMX-2P500. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a combined fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT indicator light on the front panel of the controller lights, the FAULT relay opens and a signal sound.
CONTROLS	

Error types

DEVICE CONNECTED	CAN connection between Controller and Amplifier broken.
MEMORY	Memory error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
PROTECT	Protect mode of amplifier activated.
OVERHEAT	Temperature overload of the device.
POWER SUPPLY MAINS	Error in the mains power supply of the device.
POWER SUPPLY BATTERY	Error in the battery power supply of the device.
OUTPUT PILOT DETECTION	Missing pilot tone at the amplifier output channel 1 or 2.
INPUT x PILOT DETECTION	Missing pilot tone at the amplifier input channels 1 to 4.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

PAVIRO

PVA-4CR12

The PVA-4CR12 Controller is the central paging manager for the PAVIRO system. Eight local audio inputs can be switched to four audio outputs. A two channel message manager is integrated. The controller provides all the audio processing, supervision and control functions for a complete PAVIRO system. A single controller supports up to 16 call stations and 492 paging zones. The controller is equipped with 12 zones, 18 GPIs and 19 GPOs. One controller can handle up to 2000 W loudspeaker load. Additional zones and power can be added by using up to 20 external routers and 40 amplifiers with each 2 × 500 W. The zone indicator lights on the front indicate the current status of every zone:

– Green: Zone in use for non emergency purpose
– Red: Zone in use for emergency purpose
– Yellow: Zone fault detected
– Off: Zone in idle condition

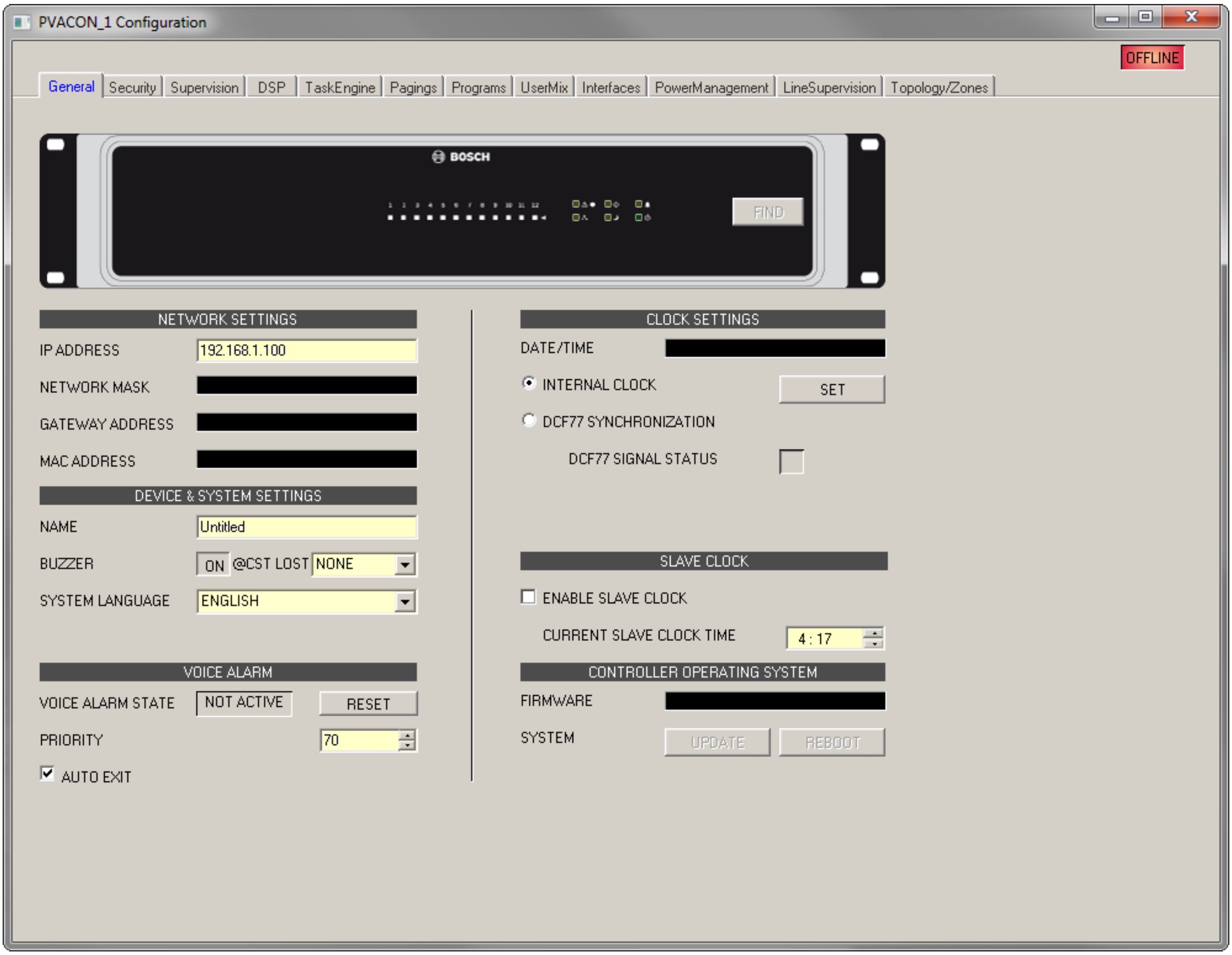
PVA-4CR12 Device

Start by creating a Controller Device in your IRIS-Net project. Drag a Controller from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu).
Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.
The following table lists all available device dialogs with a short description for each. For more detailed information, please refer to the appropriate chapters.

Dialog	Description
General	This window allows hardware settings to be configured, e.g. network settings, device name, system time and firmware version.
Security	This window allows editing passwords.
Supervision	This window provides an overview of the operational state and current fault status of the device.
DSP	This window allows editing the DSP configuration of the device.
Task Engine	This window lets you configure the Task Engine of the device.
Pagings	This window lets you configure dynamic add/sub zones (VAR pattern).
UserMix	This window lets you configure background music.
Interface	From this window the interfaces (e.g. CAN bus, GPIO control port) can all be configured. **HINT: Ethernet interface settings are edited under General dialog in the paragraph Network Settings.**
Power Management	From this window the power management of the device can be configured.
LineSupervision	The line supervision of the device can be controlled and supervised from this window.
Topology/Zones	This windows lets you configure topologies and zones of the system.

General Dialog

Double clicking on a PVA-4CR12 by default opens the General dialog box. Here, the user can make basic settings that are necessary for flawless operation. All elements of the displayed PVA-4CR12 front panel are active in on-line mode and correspond to the actual indicators on the unit.

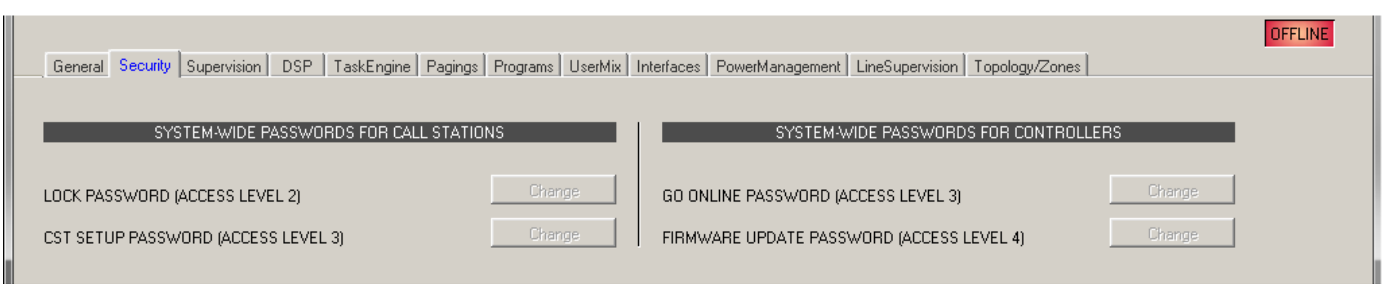


Element	Description
IP ADDRESS	Indicates the IP address of the PVA-4CR12's Ethernet port (factory setting: 192.168.1.100). Enter the address of the PVA-4CR12 with which you want to establish on-line communication.
NETWORK MASK	Indicates the Ethernet port's network mask (factory setting: 255.255.255.0).
GATEWAY ADDRESS	Indicates the standard gateway of the Ethernet port (factory setting: 192.168.1.1).
MAC ADDRESS	Indicates the MAC address of the connected PVA-4CR12 when on-line. The MAC address of the PVA-4CR12 is also shown on a label on the unit's rear panel.
NAME	IRIS-Net internal device name of the PVA-4CR12.

BUZZER	Select ON to indicate a connection failure to a call station (selectable via the drop down field) via the integrated buzzer of the PVA-4CR12.
SYSTEM LANGUAGE	Select the system language of the PAVIRO system.
VOICE ALARM STATE	This indicator shows “ACTIVE” if the device is in voice alarm state, else “NOT ACTIVE”.
RESET	Press the RESET button to deactivate the voice alarm state.
PRIORITY	Select the priority (70–100) of the voice alarm. Select OFF to disable the voice alarm handling of the device.
AUTO EXIT	Select this checkbox if the voice alarm state should be stopped automatically after the alarm signal is stopped/muted (e.g. no alarm request present).
DATE/TIME	Date and time of the PVA-4CR12 system clock.
INTERNAL CLOCK 	Opens the system clock settings dialog box.
DCF77 SYNCHRONIZATION	Select this option to synchronize the internal clock of the PVA-4CR12 with the DCF77 signal. Please refer to the manual how to connect an external DCF77 receiver.
DCF77 SIGNAL STATUS	Indicates the DCF77 signal strength: – Green: Signal strength OK – Red: Signal strength not OK
ENABLE SLAVE CLOCK	Select this checkbox if slave clocks are connected to the PVA-4CR12.
CURRENT SLAVE CLOCK TIME	Set the time for the slave clocks.
FIRMWARE	Indicates the firmware version of the PVA-4CR12 when on-line.
	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
	Reboots the PVA-4CR12.

Security Dialog

In this dialog the password of the devices can be edited.

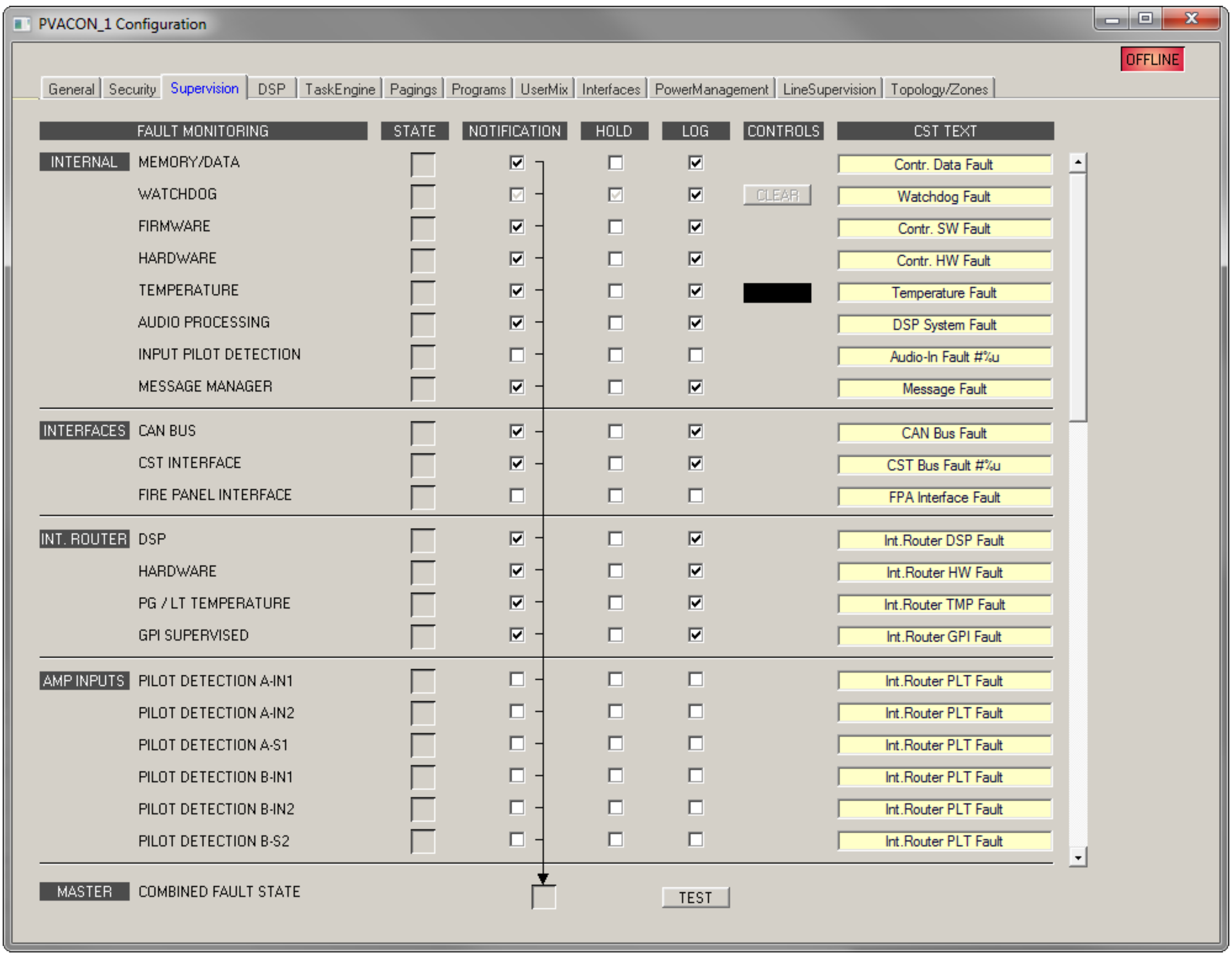


Element	**Description**
LOCK PASSWORD (ACCESS LEVEL 2)	Press the change button to edit the setting of the password for locking call stations.

CST SETUP PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for th setup of call stations.
GO ONLINE PASSWORD (ACCESS LEVEL 3)	Press the change button to edit the setting of the password for going online in IRIS-Net.
FIRMWARE UPDATE PASSWORD (ACCESS LEVEL 4)	Press the change button to edit the setting of the password for updating the firmware of the system.

Supervision Dialog

The Supervision window shows the condition of the PMX-4CR12. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT-LED on the front panel of the device lights, the FAULT relay opens and a signal sound.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the corresponding HOLD checkbox is unchecked.
LOG	
CONTROLS	
CST TEXT	If call stations are configured for error indication, the text entered here is indicated in the call station display if the error occurs. **HINT: The meaning of the parameter %u is described at the error types below.**

INTERNAL

MEMORY/DATA	Memory or Read/Write error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The device firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
HARDWARE	Error in the power supply or the A/D converters of the PMX-4CR12.
TEMPERATURE	Temperature overload of the PMX-4CR12.
Temperature control	Current temperature on the inside of the enclosure.
AUDIO PROCESSING	Error during the processing of audio data.
MESSAGE MANAGER	Error in the message manager.

INTERFACES

CAN BUS	Fault condition on the CAN bus. Further details are provided in the Interface dialog.
CST INTERFACE	Fault condition on the PCA bus. Further details are provided in the Interface dialog. The parameter %u gives the slot number of the erroneous module.
FIRE PANEL INTERFACE	Fault condition of the fire panel interface (FPA 5000).

INT. ROUTER

DSP	Error in the digital signal processing (DSP) of the device.
HARDWARE	Hardware error.

PG / LT TEMPERATURE	Temperature overload of the device.
GPI SUPERVISED	Voltage at supervised GPI out of range.

AMP INPUTS

PILOT DETECTION x-IN1	Missing pilot tone at input 1 of cluster A or B.
PILOT DETECTION x-IN2	Missing pilot tone at input 2 of cluster A or B.
PILOT DETECTION A-S1	Missing pilot tone at spare input 1 of cluster A.
PILOT DETECTION B-S2	Missing pilot tone at spare input 2 of cluster B.

EXTERNAL

CALL STATIONs	A connected DPC call station has transferred an error message. The parameter %u gives the address of the erroneous call station.
AMPLIFERS	A connected DPA power amplifier has transferred an error message. The parameter %u gives the address of the erroneous amplifier.
ROUTERS	A connected DCS system has transferred an error message. The parameter %u gives the address of the erroneous DCS system.
POWER SUPPLY	Fault condition in the power supply of the PMX-4CR12.
SPEAKER LINE FAULT	Fault condition in the speaker line supervision. The parameter %u gives the number of the erroneous speaker line, the number has following meaning: 1 to 500: Zone A 501 to 1000: Zone B

USER

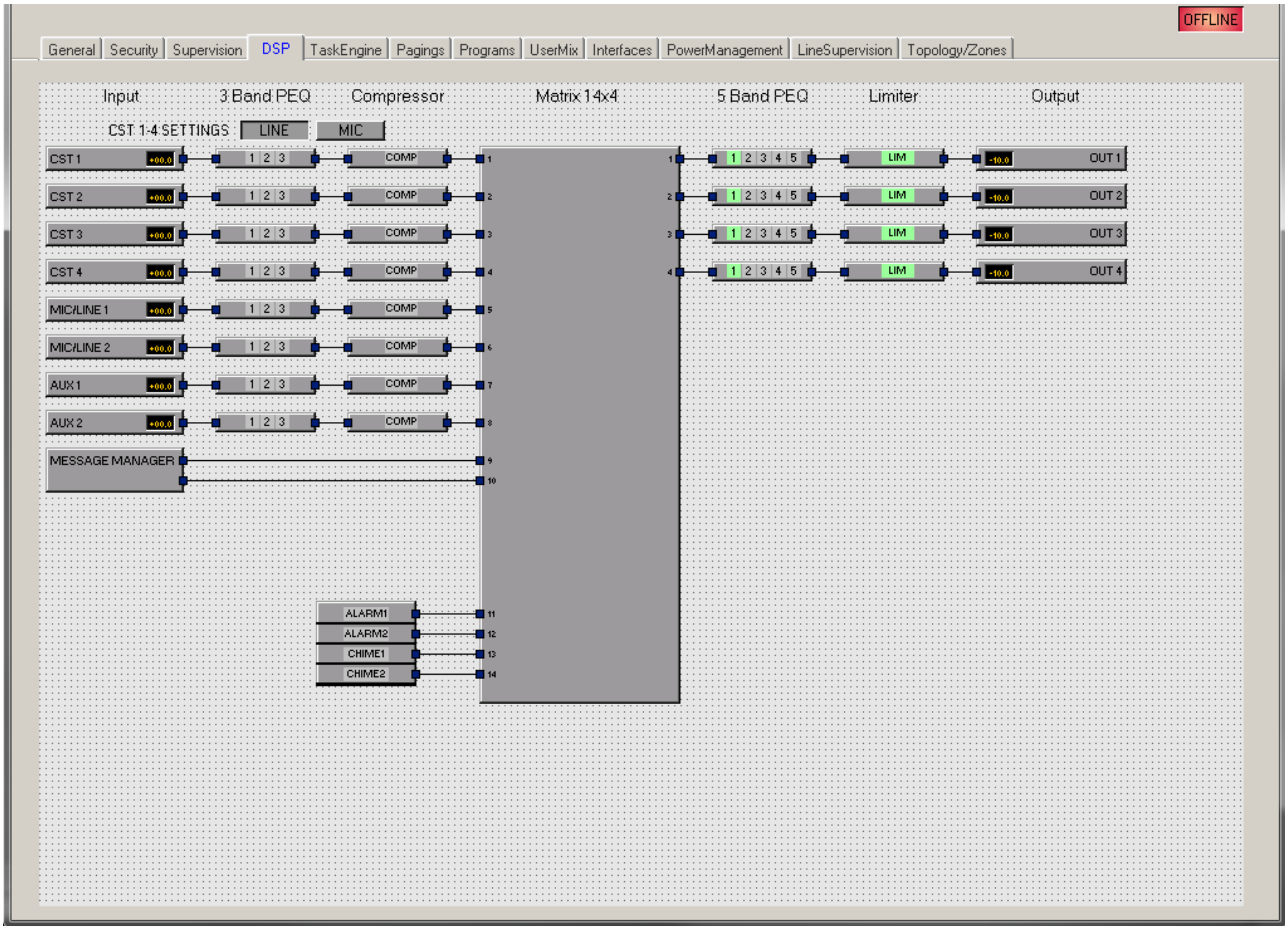
USER FAULT 1 to 10	One or more USER FAULTS have been set **HINT: Use the device Task Engine to configure USER FAULTS.**

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
	Manually setting or resetting an error.

DSP Dialog

In this dialog the DSP configuration of the Controller is shown. Double clicking on a DSP-Block's icon allows editing its configuration and settings in detail.



Input

The Input block provides access to the audio inputs of the device. The name and gain values of the input channels are indicated in the block. Double click the block to open the Inputs dialog.

Element	**Default**	**Range**	**Description**
CST 1 to 4; MIC/LINE 1,2; AUX 1,2			Permanent channel labeling.
CAN TERM/ STATE			Press the OFF button to activate the internal CAN termination resistor of the corresponding CST bus. The digit next to the button indicates the total number of termination resistors activated. The number must always be “2”.
	0.0 dB	0 to 60 dB	The gain of the MIC/LINE input channels can be adjusted in 6 dB steps.
			The +48V button of the MIC/LINE input channels is for activating phantom power whenever a suitable condenser microphone is being used.

	0.0 dB	-80 to +18.0 dB	Fader for setting the input level.
	0.0 dB	-80 to +18.0 dB	The fader display shows the numerical value of the current fader setting and additionally allows entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) pilot tone detection. The PLT button lights red when pilot tone detection is active but without a pilot signal being detected. With a pilot signal present, the PLT button lights green.
			MUTE button for muting the input signal.
LINE/MIC			Press the LINE button if audio sources with line level signals are connected to the CST busses. Press the MIC button if microphone signals are connected.

MESSAGE MANAGER
The Message Manager block provides access to the messages of the internal Message Manager. Double click the block to open the Message Manager dialog.

Element	**Description**
MESSAGES	
Active	Indicates the messages that are currently active (marked with a “X”).
Description	The unique name or description of the uploaded message. Use the corresponding text field to edit the description. The description can be edited in offline or online mode.
Type	Available message types are EVAC, Chime or Business. The Type can be set when adding messages.
Duration	The duration of the uploaded message, given in format “minutes:seconds”.
Level	Indicates the level of the message. Level ranges from -80 dB to +18 dB. Default level is 0.0 dB. Use the corresponding spin control to edit the level. The level can be edited in offline or online mode.
Info	The memory usage is indicated for all MM-2 modules.
ADD	Press the ADD button to upload a new message. A file selection dialog appears (see screenshot below) that allows selecting a message in WAV file format (mono, 48 kHz). You have to assign a description and a message type (EVAC; Chime or Business) to the message before uploading. If there are two MM-2 modules available, the location for the message has to be selected. **HINT: A selection of standard evacuation messages in different languages is available in the Download area at www.dynacord.com**
DELETE	Press the DELETE button to delete the message selected in the message list.

REPLACE	Press the REPLACE button to replace the message selected in the message list, the message type and location can not be changed. In online mode only business messages can be replaced.
ERROR STATES	
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
DETECT	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR flag is set at the same time. Additionally the FAULT-LED on the front panel of the DEVICE lights, the FAULT relay opens and a signal sounds.
MODULE	Hardware or configuration error in the MM-2 module.
MESSAGE STORAGE	Error during message storage.
PLAYBACK MEMORY	Error in the playback memory.
WATCHDOG	Watchdog error of the DEVICE. This error type is logged conforming to standards.
TEMPERATURE	Temperature of module is too high.
COLLECTED ERROR	The FAULT-LED on the front panel of the PMX-4CR12 lights at the occurrence of this type of error.
FALLBACK SIGNALS	
Fallback Evac	Select the default evacuation signal to use if no message is uploaded to the MM-2 module. This settings is valid for all PMX-4CR12 in the PROMATRIX system.
Fallback Pre-/ Chime	Select the default chime or pre-chime signal to use if no chime is uploaded to the MM-2 module. This settings is valid for all PMX-4CR12 in the PROMATRIX system.

HINT: For creating audio messages the software Audacity from http://audacity.sourceforge.net/ **can be used.**

3 BAND PEQ
Equalizers accentuate or lower the audio signal within specific frequency ranges. Eight parametric 3-Band equalizers are available.

Element	**Default**	**Range**	**Description**
CST 1-4, MIC/LINE 1-2, AUX 1-2			Press the input channel button to view or edit the corresponding PEQ settings.
LINE/MIC			Press the LINE button if audio sources with line level signals are connected to the CST busses. Press the MIC button if microphone signals are connected.
BYPASS ALL			Pressing BYPASS ALL switches off all filters.

EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope.
GAIN +0.0 dB	0 dB	-18 to +12 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
FREQ 30.0 Hz	125 Hz, 1 kHz, 16 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.1 to 100 (PEQ) 0.1 to 2.0 (Hi-/ Lopass)	Q defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi and Lo pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

COMPRESSOR

The compressor reduces the dynamic range of audio signals. Once the signal exceeds a certain threshold, the signal gets compressed, i.e. major input level changes result in minor output level changes. Narrowing the dynamic range often allows for easier recording or mixing the audio signal. Eight compressors are available.

Element	Default	Range	Description
CST 1-4, MIC/LINE 1-2, AUX 1-2			Press the input channel button to view or edit the corresponding Compressor settings.

THRESHOLD +0.0 dBu 0.775 V	+6.0 dBu or 1.546 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	THRESHOLD defines the signal level at which the Compressor sets in. Entering the desired value is possible in dBu as well as in V. The entered value is automatically converted in both directions.
RATIO 1.0:1	4.0:1	1.0:1 to 8.0:1	RATIO defines the compression rate, i.e. the degree of compression above the threshold level. For example, a rate of 4.0 : 1 represents a signal reduction by factor 4.
ATTACK 5 ms	5 ms	0 to 99 ms	ATTACK defines the velocity, at which the compressor sets in. A short attack rate means that even short signal peaks are efficiently com- pressed. Longer attack rate leave signal peaks untouched.
RELEASE 250 ms	250 ms	0 to 999 ms	RELEASE defines the control time interval the compressor takes to return to an uncompressed signal level, after the signal dropped below the set threshold.
BYPASS			BYPASS activates (not engaged) or deactivates the Compressor (engaged), which allows for quick A / B comparison between the com- pressed and uncompressed audio signal.

5 BAND PEQ
Equalizers accentuate or lower the audio signal within specific frequency ranges. Four parametric 5-Band equalizers are available.

Element	Default	Range	Description
OUT 1-4			Press the output channel button to view or edit the corresponding Parametric EQ settings.
BYPASS ALL			Pressing BYPASS ALL switches of all filters.
EQ1			Name of the corresponding filter band. Clicking with the right mouse button onto this field opens Copy & Paste menu, which allows comfortably copying all EQ parameters of the selected filter to any other EQ within the same project.
TYPE PEQ	PEQ	PEQ. Loshelv. Hishelv, Hipass, Lopass	TYPE defines the filter type. – PEQ is a parametric Peak Dip Filter with its frequency, quality (Q) and gain being programmable. – Loshelv / Hishelv create a Low-Shelving or High-Shelving filter with the parameters being: frequency, slope and gain. – Lopass / Hipass creates a Low Pass or High Pass filter with adjustable frequency and slope.

GAIN +0.0 dB	0 dB	-18 to +18 dB	GAIN defines the amplification (increase) or attenuation (reduction) of parametric EQs or low shelving and high shelving equalizers.
FREQ 30.0 Hz	60 Hz, 250 Hz, 1 kHz, 4 kHz, 19 kHz	20 Hz to 20 kHz	FREQ (frequency) sets the center frequency of a parametric EQ or the cut-off frequency of shelving and Hi / Lo pass filters.
Q 0.7	0.7	0.01 to 6.67 Oct. or 0.1 to 40 (PEQ) 0.1 to 2.0 (Hi-/ Lopass)	Q or BW defines the quality or bandwidth of a parametric EQ. A high Q-value results in a narrowband filter, while a small Q-value results in a broadband filter. The Q-value also sets the quality and thus the response of Hi and Lo pass filters with slopes of 12dB/ oct.
SLOPE 6dB/Oct	6dB/Oct	6dB/Oct, 12dB/Oct	SLOPE sets the steepness or filter-order of low or high shelving equalizers and low or high pass filters. Setting different slopes within the transmission range is possible. That, in conjunction with the Q-parameter, offers the possibility for a hi-pass filter to be programmed for B6-alignment, which describes a drastic rise in the cut-off frequency range.
BYPASS			BYPASS switches the corresponding filter ON (not engaged) or OFF (engaged), which allows for quick A / B-evaluation of the actual effect that a filter has on the sound.

LIMITER

A Limiter is used when the output signal must not exceed a specific peak level, independent of how much the input level rises. Short attack times effectively limit overshoots. Limiters are often used as protection for the components following them an audio chain, i.e. to prevent an amplifier from clipping or protect loudspeaker systems against mechanical damage.

Element	**Default**	**Range**	**Description**
OUT 1-4			Press the output channel button to view or edit the corresponding Peak Limiter settings.
THRESHOLD +0.0 dBu 0.775 V	+6.0 dBu or 1.546 V	-9.0 to +21.0 dBu or 0.275 to 8.696 V	The THRESHOLD parameter defines the level value at which the limiter sets in. Signal levels below the threshold will pass through the limiter unaffected. As soon as the signal level reaches or exceeds the threshold, signal limiting sets in. Entering the threshold value is possible in dBu or V. The value can be entered in either box and will automatically be converted in the other.
ATTACK 5 ms	5 ms	0 to 50 ms	ATTACK defines how fast the gain is reduced after the signal exceeds the threshold level.

	100 ms	10 to 1000 ms	RELEASE defines how fast the output signal returns to its normal level once it drops below the threshold.
			BYPASS activates (not engaged) or deactivates (engaged) the Limiter, which allows for quick A / B comparison between the limited and unlimited audio signal.

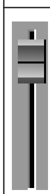
OUTPUT

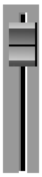
The Output block provides access to the outputs of the device. The name and gain values of the out channels are indicated in the block. Double click the block to open the Output dialog.

Element	Default	Range	Description
OUT 1-4			Permanent channel labeling.
	0.0 dB	-80 to +18.0 dB	Fader for setting the output level.
	-10.0 dB	-60 to +6 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
			The PLT button activates (engaged) or deactivates (not engaged) the pilot tone generator.
			MUTE button for muting the output signal.

ALARM CHIME

The Alarm Chime dialog allows the configuration of the internal alarm and chime generators.

Element	Default	Range	Description
Alarm Configuration			
	-3.0 dB	-80 to 0 dB	Fader for setting the alarm level.
	-3.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
NEW ALARM			Press this button to ad a new alarm to the alarm list.
PLAY ALARM			Press this button to playback the alarm selected in the alarm list.

Chime Configuration			
	-9.0 dB	-80 to 0 dB	Fader for setting the chime level.
	-9.0 dB	-80 to 0 dB	The fader display shows the numerical value of the current fader setting and additionally provides the possibility for entering a desired value.
PLAY CHIME			Press this button to playback the chime selected in the chime list.

MATRIX

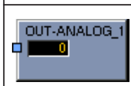
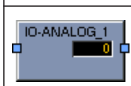
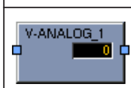
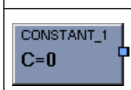
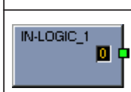
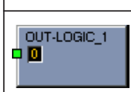
Double click on the Matrix 14x4 to open the Matrix 10x4 dialog (the 4 missing inputs in this dialog are used for the internal generators of the PMX-4CR12). The Matrix 10x4 allows connecting inputs and outputs. Left clicking the node in the matrix where the output channel's column and the input channel's line meet with the mouse does connect an output to an input. Clicking again onto the corresponding node disconnects inputs and outputs.
Please note following restrictions for making connections in the matrix:
- BGM inputs can only routed via a call station, so this is not possible in this dialog
- Unused inputs can not be routed
- Inputs used for alarms, announcements etc. can not be routed
- Inputs used for the Message Manager can not be routed
- Manual routings override existing BGM routings

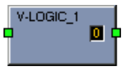
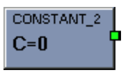
Element	**Default**	**Range**	**Description**
DUCKING	-40 dB	-85 to 0 dB	The signal level of the back ground music is reduced by the level entered here when the in input signal, signal level reaches or exceeds a set threshold.
FADE IN	0.02 s	0.01 to 4 s	FADE IN defines how fast the gain of the background music signal is reduced after the input signal exceeds the threshold level.
FADE OUT	0.02 s	0.01 to 0.4 s	FADE OUT defines how fast the gain of the background music signal is returned to the preset level once the input signal drops below the threshold level.

TaskEngine Dialog

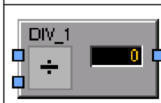
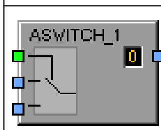
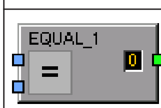
The Task Engine Window allows configuring the Task Engine by dragging inputs, links or outputs from the categories of the FUNCTIONS AND IOS on the left corner of the screen into the Task Engine Worksheet. Elements can be freely positioned and wired within the worksheet. Double clicking on inputs or outputs allows configuring them in detail. Copy & paste of blocks allow convenient editing of the Task Engine configurations. The size of the worksheet automatically increases when a block is moved to the current border.
Configuring the Task Engine as well as wiring DSP blocks is possible only in offline mode. Please refer to section "How to configure a Control" on page 20 how to assign functions and connections to a Task Engine block.

VALUES

Element	Description
	The Input Analog block is a variable parameter for rational numbers which always outputs the current value of the connection.
	The Output Analog block is a variable parameter for rational numbers which always assigns the current value at the input to the connection.
	The IO Analog block is a variable parameter for rational numbers which always outputs the current value of the connection. The current value at the input is assigned to the connection.
	The Virtual Analog Value block is similar to the IO Analog block, but has no connection. The keyword Value of the block is used instead of a connection. The keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the DPM. – Persistent = 0: The value is stored in volatile memory.
	The Constant Analog block is a constant parameter for rational numbers. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Input Logic is a variable parameter for Boolean values which always outputs the current value of the connection.
	The Output Logic block is a variable parameter for Boolean values which always assigns the current value at the input to the connection.

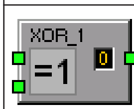
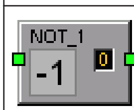
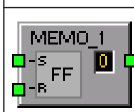
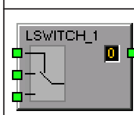
	The IO Logic block is a variable parameter for Boolean values which always outputs the current value of the connection. The current value at the input is assigned to the connection.
	The Virtual Logic block is similar to the IO Logic block, but has no connection. The keyword Value of the block is used instead of a connection. he keyword Persistent is used for persistent storage of the value: – Persistent = 1: The value is stored in non-volatile memory, so the value is still available after a reset of the device. – Persistent = 0: The value is stored in volatile memory.
	The Constant Logic block is a constant parameter for Boolean values. The block always outputs the value assigned to its keyword Value during Task Engine configuration.
	The Calendar Entry block is being used to create time-dependant Boolean values. The outputted Boolean value depends on the configuration of this block and the current system time.

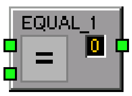
ANALOG OPERATIONS

Element	Description
	The Addition block provides 2 inputs for rational numbers. The rational number at the output is always the sum of rational numbers of the (wired) inputs.
	The Subtraction block subtracts the rational number of the lower input from the rational number of the upper input. The output always presents the result of this analog operation.
	The Multiplication block multiplies the rational number of the upper input with the rational number of the lower input. The output always presents the result of this analog operation.
	The Division block divides the rational number of the upper input by the rational number of the lower input. **CAUTION: If the rational number "0" is present at the lower input, the rational number "0" is always output, independent of the upper input's value.**
	The Switch block switches the rational number at the center or lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Convert block converts a Boolean value to a rational number. The Boolean value 0 is converted to the rational number 0.0, the Boolean value 1 is converted to the rational number 1.0.
	The Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when identical numbers are present at the inputs.
	The Not Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the numbers that are present at the inputs differ from each other.

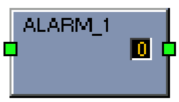
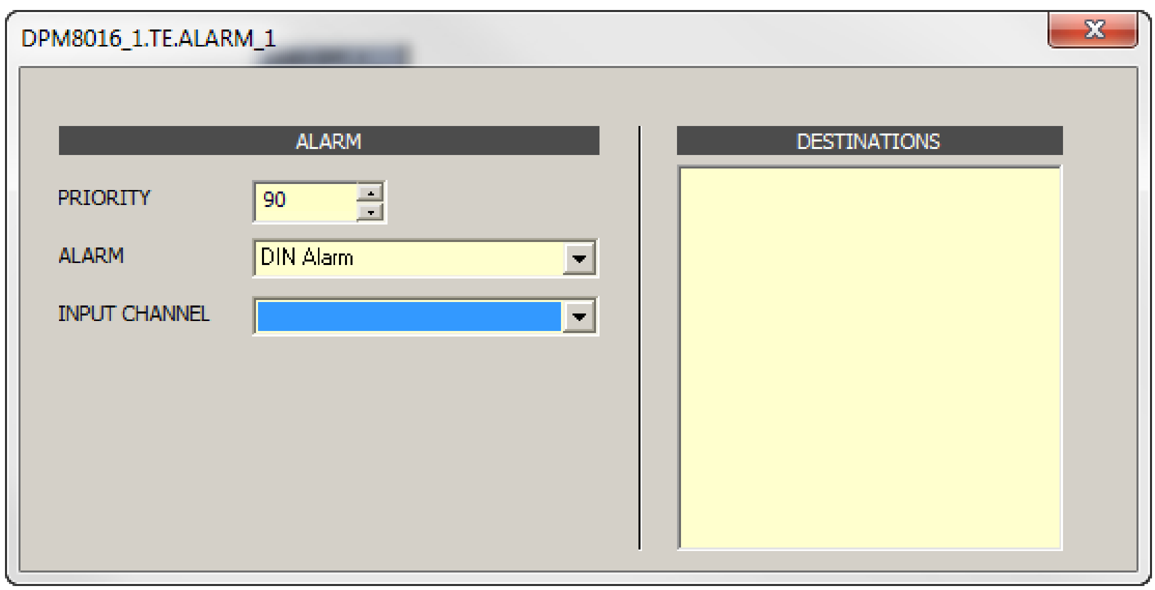
	The Greater block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater than the rational number of the lower input.
	The Greater Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is greater or equal than the rational number of the lower input.
	The Less block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower than the rational number of the lower input.
	The Less Or Equal block compares the rational numbers at the inputs. The Boolean value at the output is only true when the rational number of the upper input is lower or equal than the rational number of the lower input.
	The Range Check block allows the range check of a rational number. The Boolean value at the output is only true when the rational number at the input is greater or equal than the property TE.RANGE_CHECK_1.range.min **and** lower or equal than the property TE.RANGE_CHECK_1.range.max – range.max: Enter the upper bound of the range to be checked – range.min: Enter the lower bound of the range to be checked

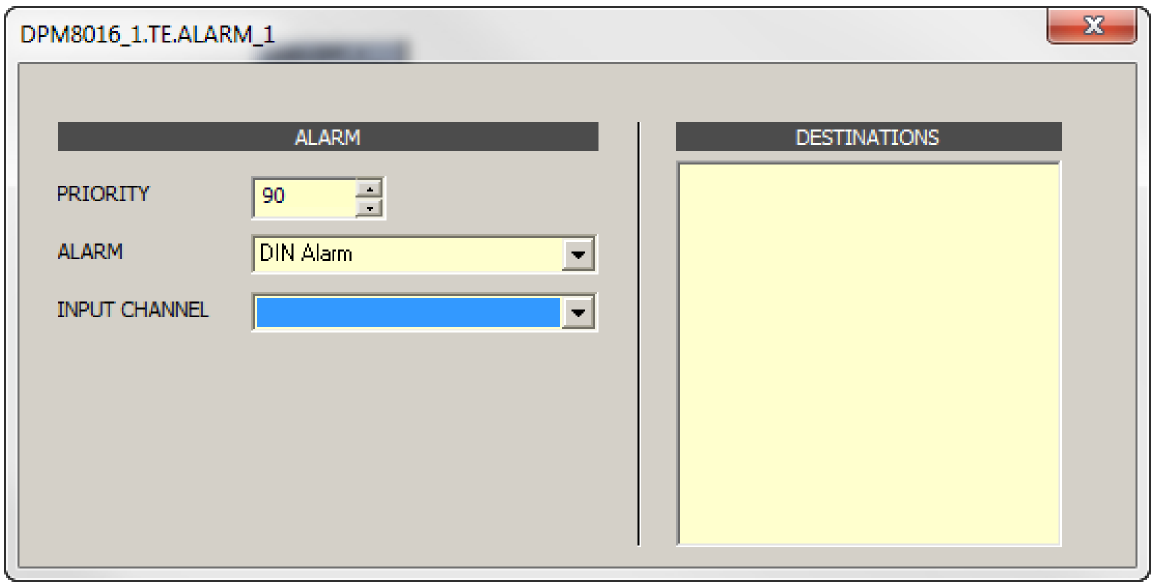
LOGIC OPERATIONS

Element	Description
	The AND block provides 2 inputs for Boolean values. The Boolean value at the output is only true when all (wired) inputs are true.
	The OR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when at least one (wired) input is true.
	The XOR block provides 2 inputs for Boolean values. The Boolean value at the output is only true when exactly one (wired) input is true.
	The NOT block negates the Boolean value at the input.
	The Memo (Flip-flop) block provides 2 inputs for Boolean values. The upper input sets the flip flop, the lower input resets the flip flop.
	The Switch block switches the Boolean value at the center or the lower input through, depending on the Boolean value at the upper input. If the Boolean value at the upper input is false, the value of the center input appears at the output. If the Boolean value at the upper input is true, the value of the lower input appears at the output.
	The Convert block converts a rational number to a Boolean value. The rational number 0.0 is converted to the Boolean value 0, the rational number 1.0 is converted to the Boolean value 1.

	The Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are identical (e.g. both inputs are true, or both inputs are false).
	The Not Equal block compares the Boolean values at the inputs. The Boolean value at the output is only true when the values at the inputs are different from each other (e.g. one input is true while the other input is false).

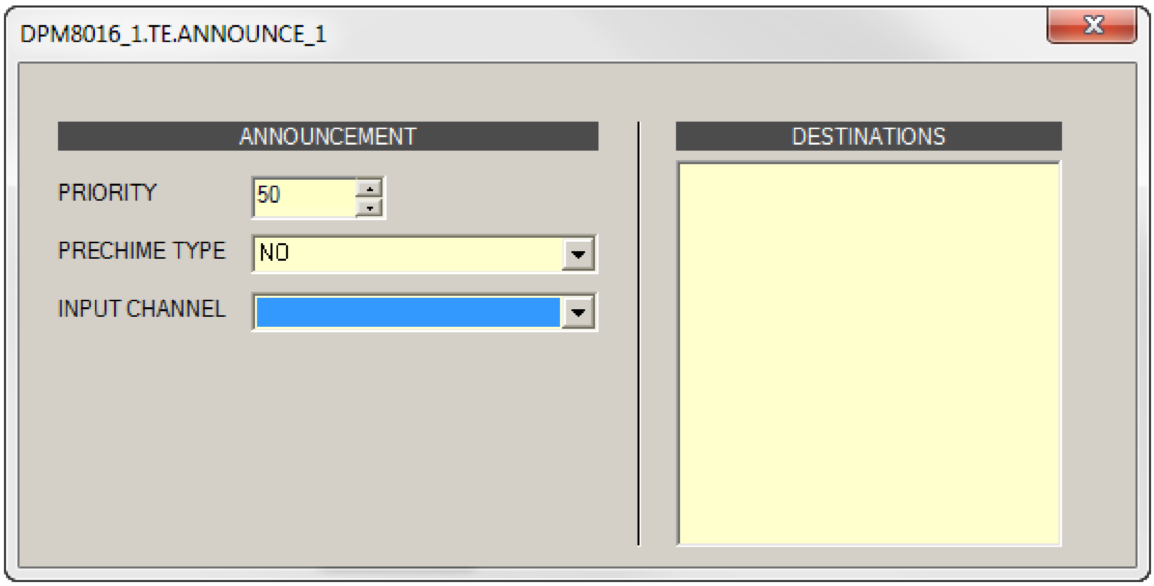
ADVANCED OPERATIONS

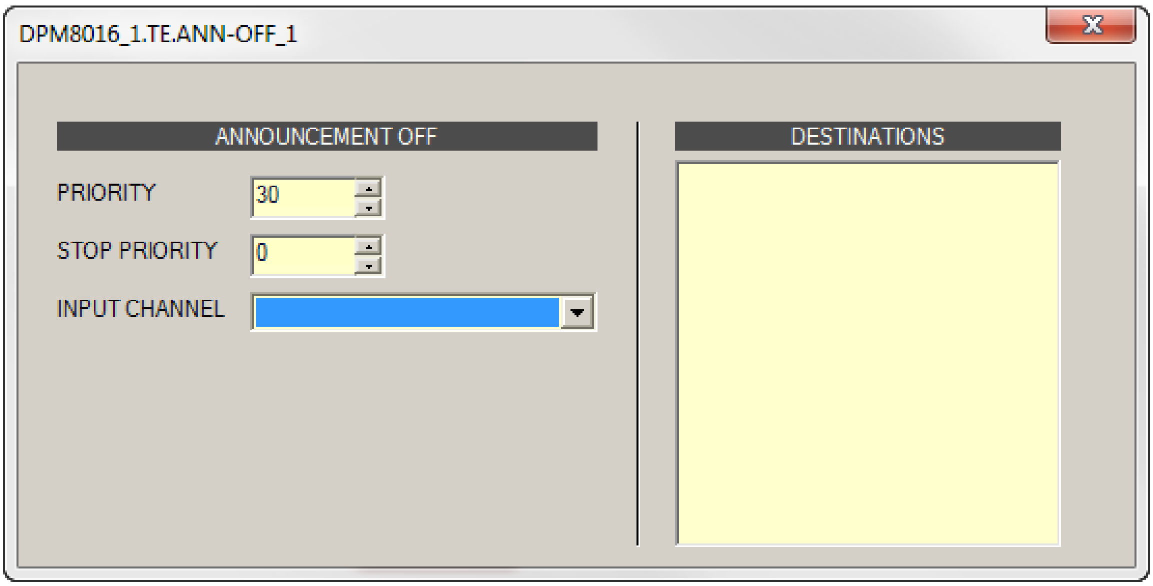
Element	Description
	The Alarm block is used to trigger an alarm. Double click the block to edit the alarm settings (see screenshot below). – PRIORITY: Priority of the alarm (0 to 100). – ALARM: Select the alarm type to be triggered, see table below. – INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the DPM 8016, where the external alarm signal is present. – DESTINATIONS: Select the destination zone(s) or group(s) for the alarm. 

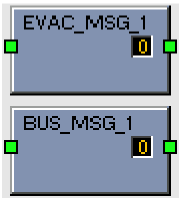
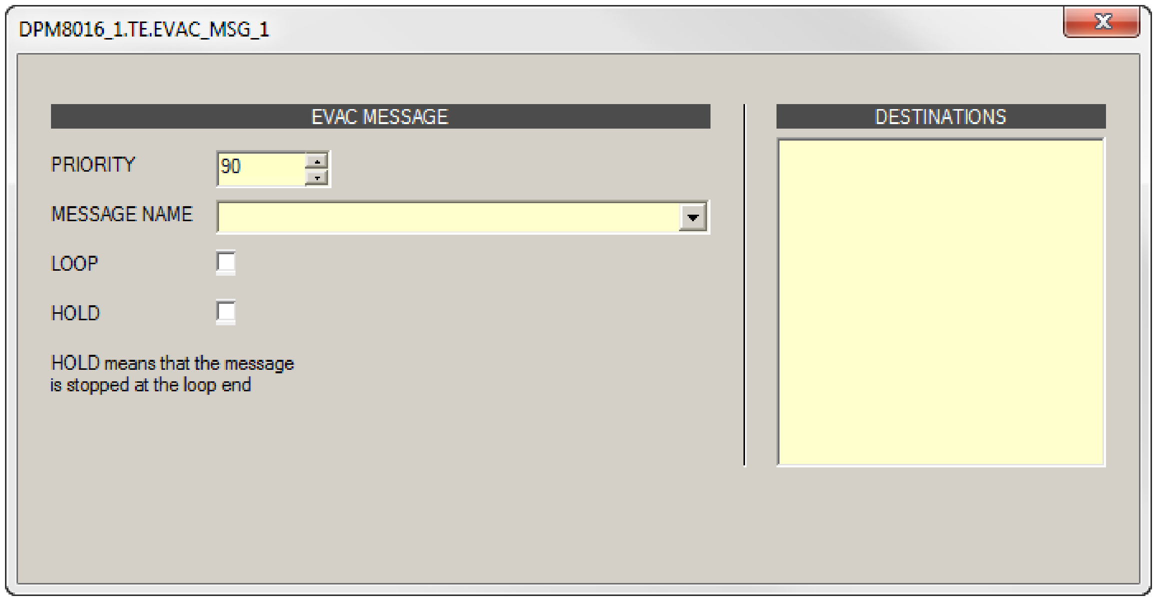
	The Manual Alarm is similar to the Alarm block. The additional input T acts like a pushbutton and allows to switch the alarm signal on or off. Double click the block to edit the alarm settings (see screenshot below). – PRIORITY: Priority of the alarm (0 to 100). – ALARM: Select the alarm type to be triggered, see table below. – INPUT CHANNEL: When using ALARM = EXTERN select the input channel of the device, where the external alarm signal is present. – DESTINATIONS: Select the destination zone(s) or group(s) for the alarm. 

<table>
<tr>
<td>
</td>
<td>The Announcement block is used to trigger an announcement. Double click the block to edit the announcement settings (see screenshot below).

PRIORITY: Priority of the announcement (0 to 100).
PRECHIME TYPE: Select the pre chime (see table below). Select NO, if there should be now pre chime
INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present.
DESTINATIONS: Select the destination zone(s) or group(s) for the announcement.

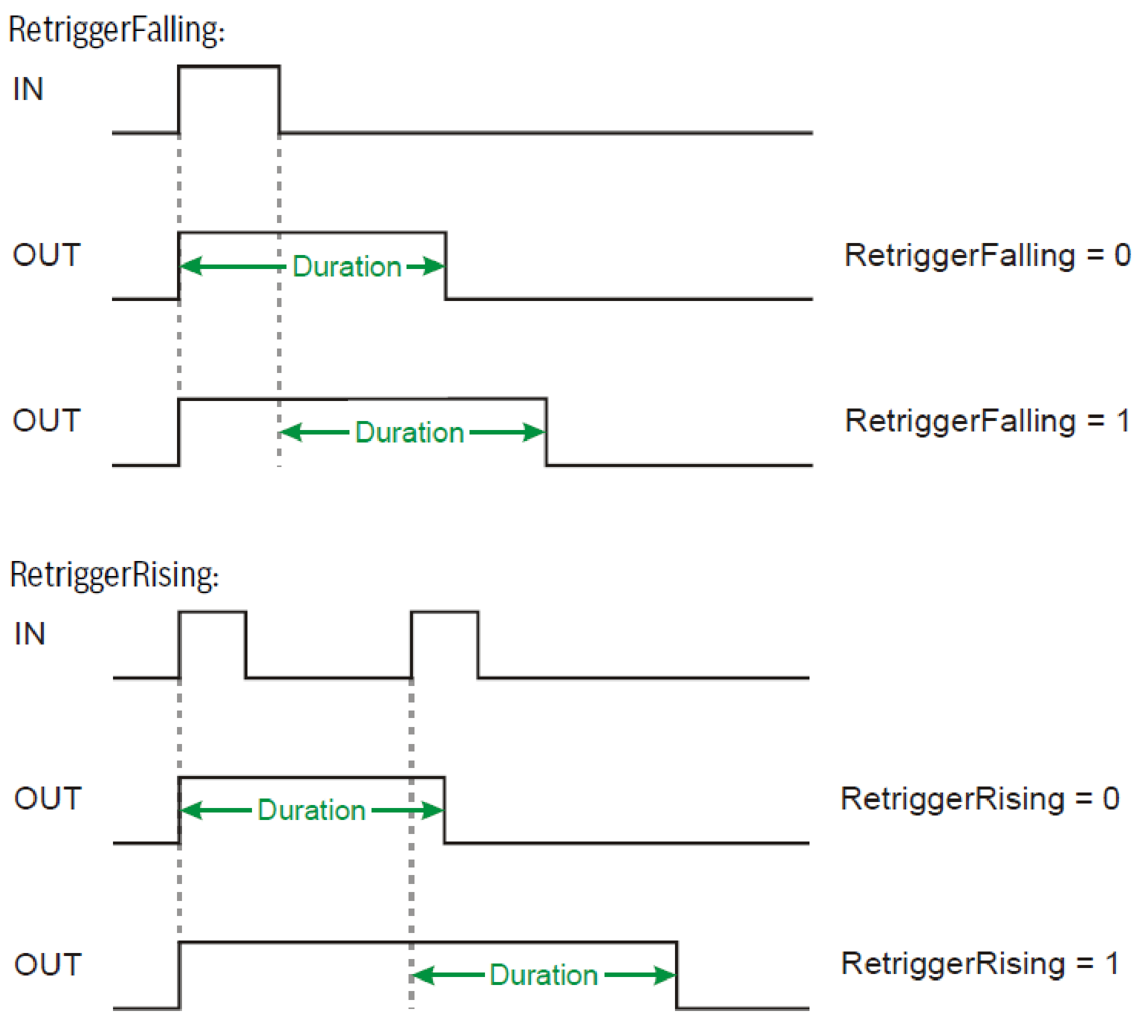

</td>
</tr>
</table>

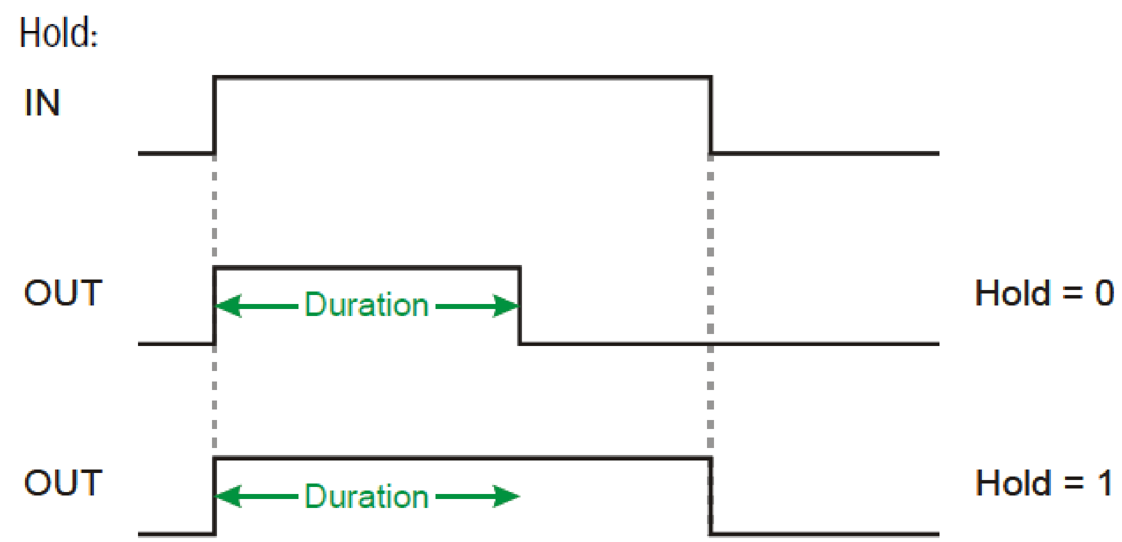
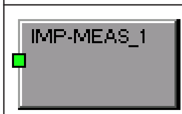
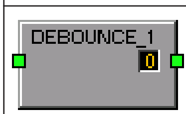
<table>
<tr>
<td>
</td>
<td>The Announcement OFF block is used to stop an announcement. Double click the block to edit the announcement settings (see screenshot below).
– PRIORITY: Priority of the announcement (0 to 100).
– STOP PRIORITY: Enter the priority (0 to 100) that is used to stop an announcement.
– INPUT CHANNEL: Select the input channel of the device, where the announcement signal is present.
– DESTINATIONS: Select the destination zone(s) or group(s), where the announcement should be stopped.

</td>
</tr>
<tr>
<td>
</td>
<td>The Chime block is used to trigger a chime. Double click the block to edit the chime settings.
– PRIORITY: The priority of the chime (0 to 100).
– TYPE: Select the type of the chime.
– LOOP: Select this checkbox to repeat the chime automatically.
– HOLD: Hold means that the message is stopped at the loop end.
– DESTINATIONS: Select the destination zone(s) or group(s) for the chime.</td>
</tr>
</table>

	The EVAC Message or Business Message block is used to trigger a MM-2 message. Double click the block to edit the message settings (see screenshot below). – PRIORITY: The priority of the message (0 to 100). – MESSAGE NAME: Select the (EVAC or Business) message to start. – LOOP: Select this checkbox to repeat the message automatically. – HOLD: Hold means that the message is stopped at the loop end. – DESTINATIONS: Select the destination zone(s) or group(s) for the message. 

The Timer block sets the state at the output to true for an adjustable duration, when the Boolean value at the input changes from false to true.

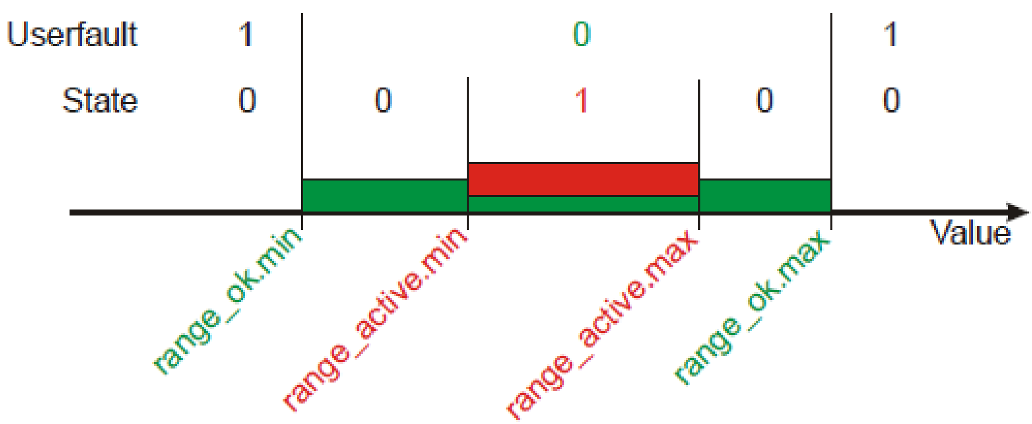
- Duration: Enter the duration in seconds, without unit.
- Hold: See illustration below.
- Retrigger Falling: See illustration below.
- Retrigger Rising: See illustration below.
- State: State of the block (1 = time running)
- Timer Value



	
	The CST Text block is used for indicating a text message at the LC display of one or more call stations. – Acknowledge: Enter 1 if pressing the ESC button at the call station should discard the text at the display. – Address: Enter the CAN address of the call station, where the text should be indicated. Enter 0 if the text should be indicated at all call stations. – Buzzer: Enter 1 if the buzzer should signal the text indication additionally. – Clear: Enter 1 if the text should be cleared when the input changes from true to false. – Duration: Enter the time in seconds (without unit), how long the text should be indicated. – State: State of the block (1 = text is indicated) – Text: Enter the text to be indicated at the display. The maximum length is 20 characters, including space and special characters. See table below for available characters.
	The Impedance Measurement block is used for executing a line measurement. – Lines By Name = ALL – State: State of the block (1 = measurement active) – Test Function = LINETEST
	The Debounce block is used for debouncing a signal. – Falling Edge: Enter 1 if the falling edge (change from true to false) at the input should be debounced. – Rising Edge: Enter 1, if the rising edge (change from false to true) at the input should be debounced. – State: State of the block – Time: Enter the debounce time in seconds (without unit)
	The Loop block allows building feedback loops in the Task Engine. Unstable conditions are prevented by the block. To point out the function of this block, the input is at the right side, the output is at the left side.
	The Text Box allows labeling the task engine configuration. Click the Modify Properties entry in the context menu to open the Edit Textbox dialog. This dialog allows editing the caption and e.g. font size and font type.

<table>
<tr>
<td>
</td>
<td>The Input Supervision block allows supervision of a rational number, especially an input signal from a CIE (Control and Indicating Equipment/fire alarm system). Two ranges can be defined, the Active range and the Ok range. Depending on the ranges the Boolean value at the output (e.g. for triggering an alarm) and a USER FAULT (e.g. for error indication of invalid input values) will be set.
The Active range is defined by:
– range_active.max: Upper bound of the Active range
– range_active.min: Lower bound of the Active range
The Boolean value at the output is true if the rational number assigned via Function & Connection is within the Active range. The Boolean value at the output is false if the rational number at the input is below or above the Active range.
The Ok range is defined by:
– range_ok.max: Upper bound of the Ok range.
– range_ok.min: Lower bound of the Ok range.

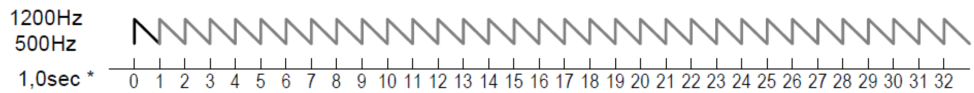
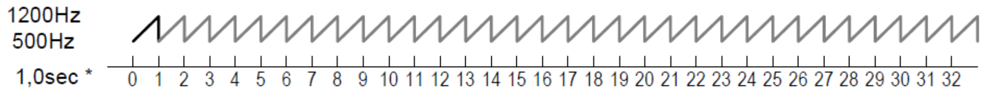
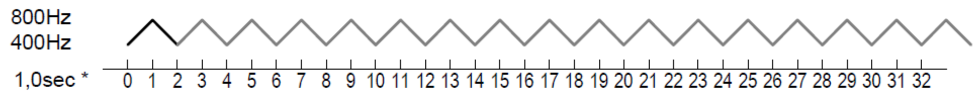
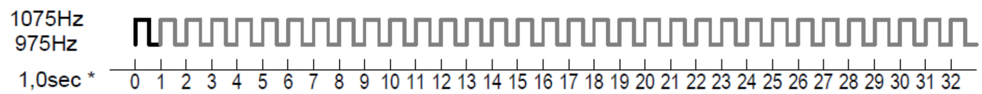
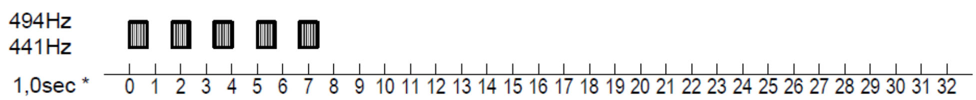
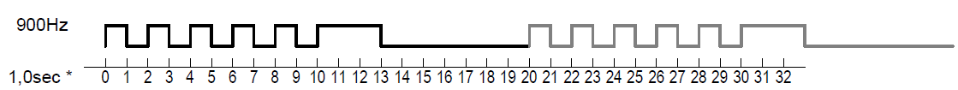
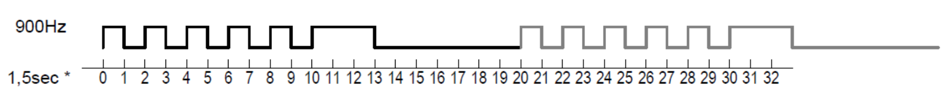
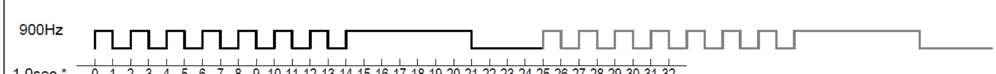
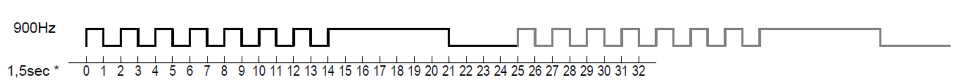
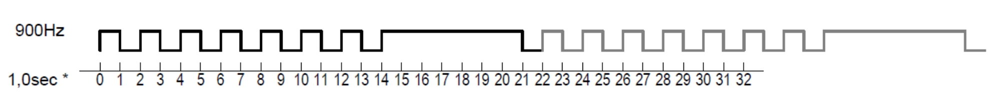
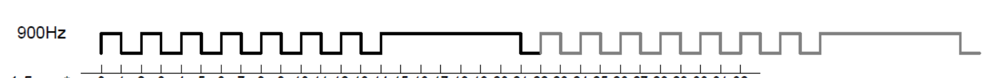
HINT: If the value of the assigned Function & Connection leaves the Ok range, the State does not change (“state value is latched“)

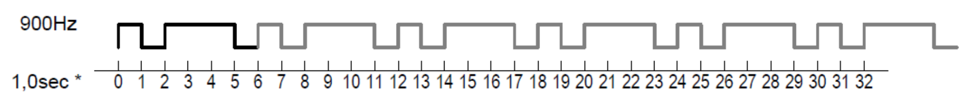
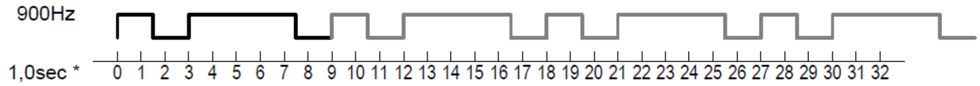
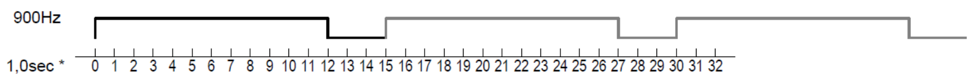
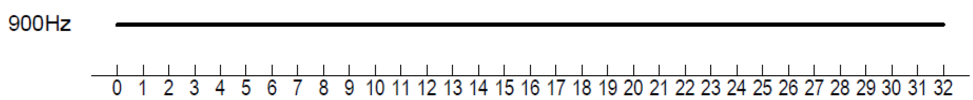
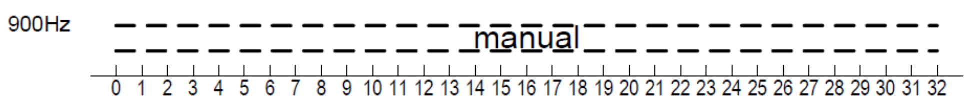
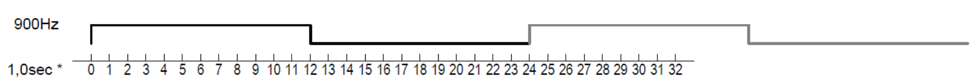
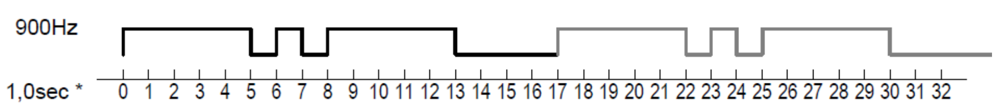
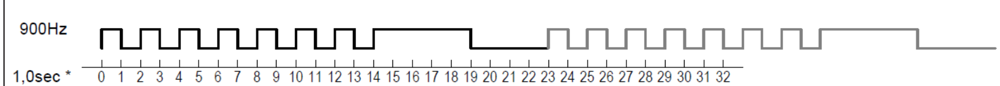
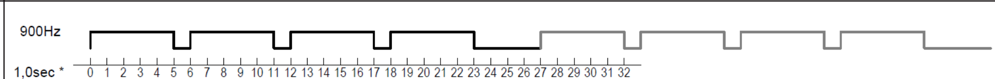
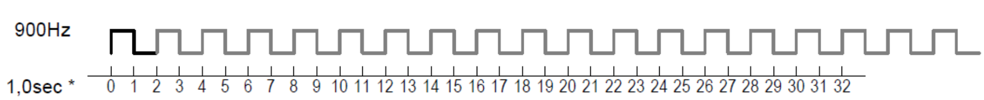
The USER FAULT is set to 0 if the rational number assigned via Function & Connection is within the Ok range. The USER FAULT is set to 1 if the rational number at the input is below or above the Ok range. Following properties are used to select the USER FAULT:
– userfault_connection
– userfault_idx

</td>
</tr>
<tr>
<td>Superblocks</td>
<td>Superblocks are listed here. Please refer to page 240 how to use Superblocks.</td>
</tr>
</table>

Chime types

Typ
1_TONE
2_TONE
3_TONE
4_TONE
2x2_TONE
2_TONE_PRE

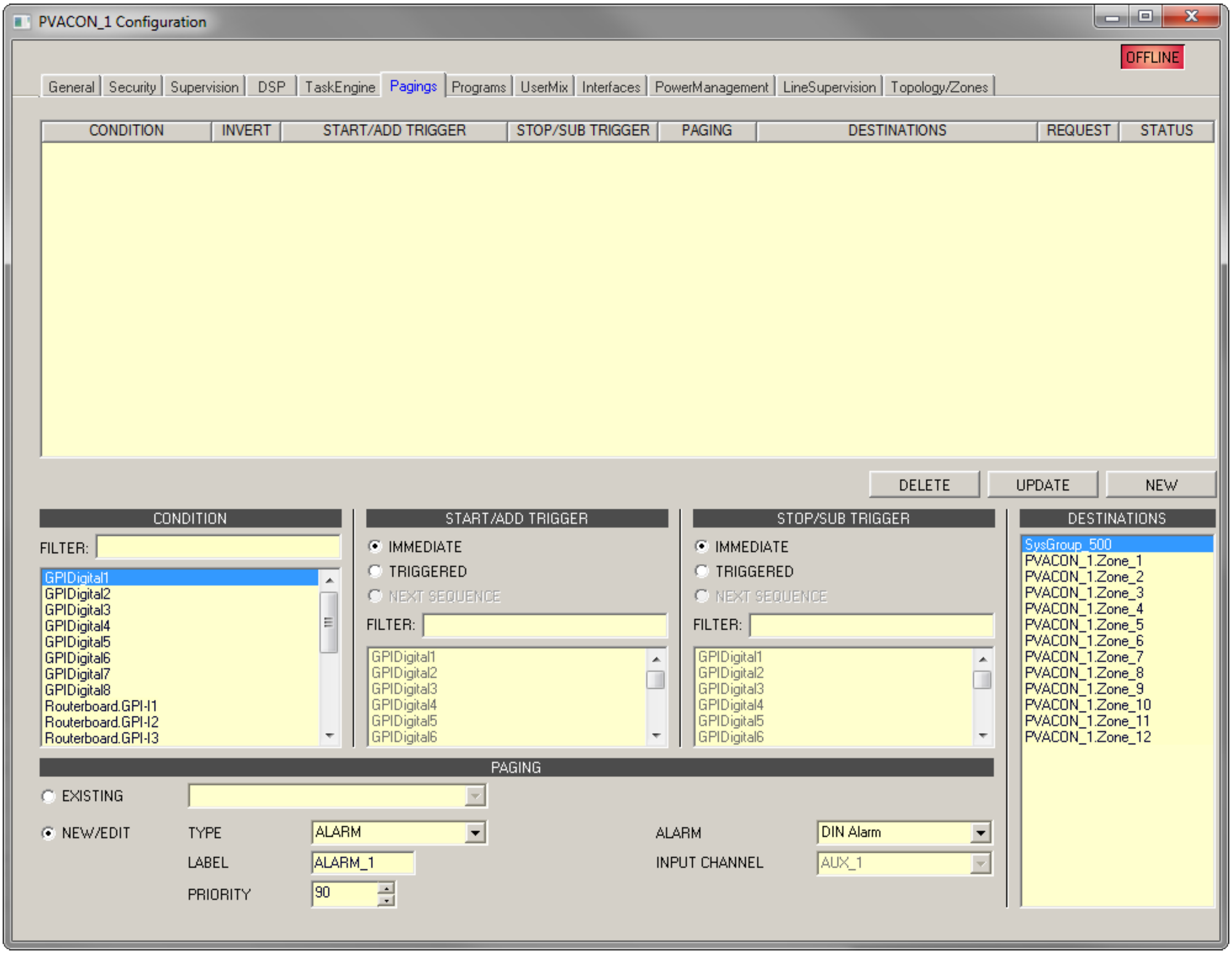
Alarm Types

Type	Graphical Illustration
Extern	-
DIN Alarm	
Slow Whoop	
Siren	
Two-Tone Alarm	
Telephone Alarm	
Ship Alarm 1	
Ship Alarm 2	
Ship Alarm 3	
Ship Alarm 4	
Ship Alarm 5	
Ship Alarm 6	
Ship Alarm 7	

Ship Alarm 8	
Ship Alarm 9	
Ship Alarm 10	
Ship Alarm 11	
Ship Alarm 12	
Ship Alarm 13	
Ship Alarm 14	
Ship Alarm 15	
Ship Alarm 16	
Ship Alarm 17	
SILENCE	-

Pagings Dialog

This dialog allows the configuration of pagings (e.g. alarm or EVAC messags) with dynamic destinations.



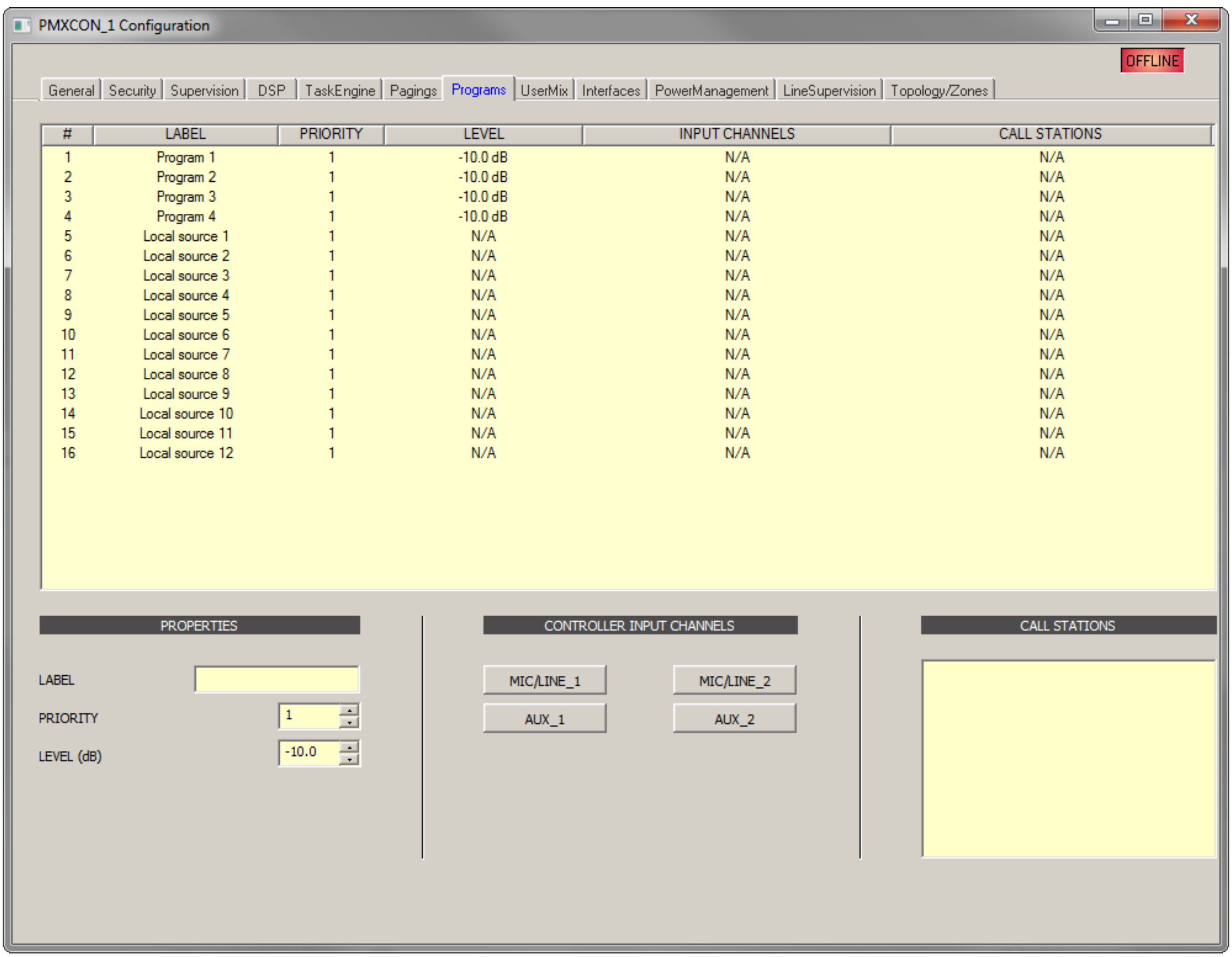
Element	Description
CONDITION	The status of the condition selected here starts the paging, e.g. the contact of a CIE connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the paging.
START/ADD TRIGGER	The value used to trigger the start of (or addition of destinations to) an active paging. The rising edge of the value is evaluated.
STOP/SUB TRIGGER	The value used to trigger the stop of (or subtraction of destinations from) an active paging. The rising edge of the value is evaluated.
PAGING	The paging initiated by the condition.
DESTINATIONS	The destinations (zones or groups) of the paging.
REQUEST	Indicates if the paging condition is active or inactive.
STATUS	Indicates if the paging is ON or OFF.

Element	Description
DELETE	Press the DELETE button to delete the paging selected in the paging list.
UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to paging selected in the paging list.
NEW	Press the NEW button to create a new paging using the settings in the lower section of the dialog and adds it to the paging list.
CONDITION	
FILTER and condition list	Select the condition to start a paging from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
START/ADD TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should start immediately or the zones should be added immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be added only after the message ended. When selected, the paging is started immediately. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
STOP/SUB TRIGGER	
IMMEDIATE	Select IMMEDIATE if the paging should stop immediately or the zones should be removed immediately.
TRIGGERED	Select TRIGGERED if the paging should be triggered by the value selected below.
NEXT SEQUENCE	Select NEXT SEQUENCE if zones should be removed only after the message ended. When selected, the paging is stopped immediately after the message ended. Can only be used for MM-2 Messages.
FILTER and trigger list	Select the trigger condition from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
PAGING	
EXISTING	Select EXISTING to select an existing paging from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the paging.
TYPE	Select the paging type from the dropdown.
LABEL	Enter the name of the paging.
PRIORITY	Select the priority of the paging.
ALARM	If the selected paging TYPE = ALARM you can select the alarm type from this dropdown.
PRECHIME TYPE	If the selected paging TYPE = ANNOUNCEMENT you can select the prechime type from this dropdown.

CHIME TYPE	If the selected paging TYPE = CHIME you can select the chime type from this dropdown.
MESSAGE NR	If the selected paging TYPE = EVAC Message you can select the message number from this dropdown.
INPUT CHANNEL	If the selected paging TYPE= ANNOUNCEMENT or TYPE = ALARM (and ALARM = Extern) you can select the audio input channel for the paging.
DESTINATIONS	Select the zones or groups of the paging.

Programs Dialog

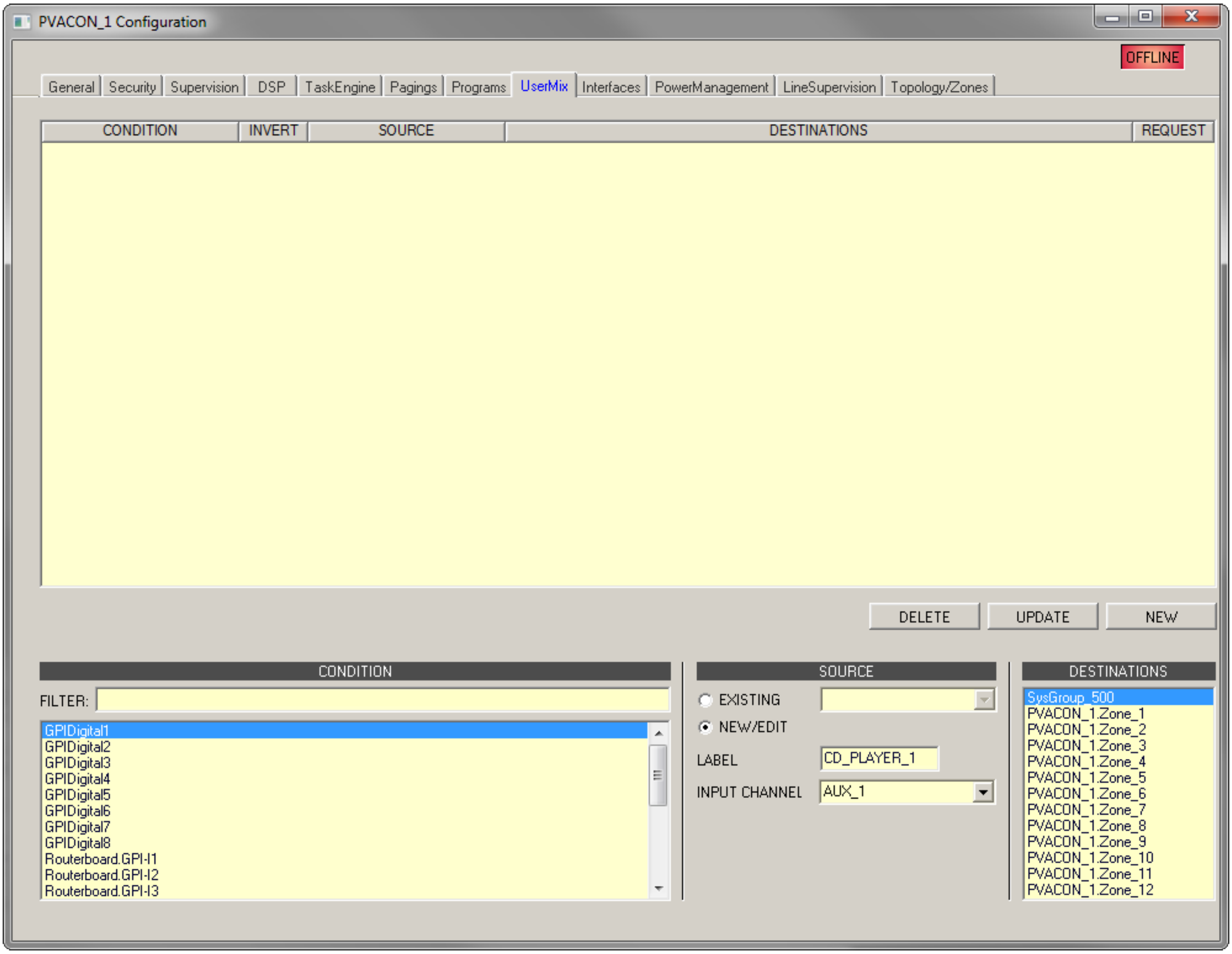
The Programs dialog allows to configure 4 programs for back ground music.



Element	Description
#	Number of the program.
LABEL	Name of the program.

Element	Description
PRIORITY	The priority of the program.
LEVEL	Level of the program.
INPUT CHANNELS	Input channel of the program. Select more than one input channel to mix the audio signals.
CALL STATIONS	The call stations where this program is listed in the menu and can be selected by the call station user.
LABEL	Text field for labeling a program (max. 20 characters), e.g. giving it an application specific name. Note: Using “,” (comma) in a name is not permissible.
PRIORITY	Edit the priority of the program selected in the program list (range: 1 to 69).
LEVEL (dB)	Edit the level of the program selected in the program list (range: -80 to 0 dB). Only the level can be edited in online mode.
CONTROLLER INPUT CHANNEL: MIC/LINE 1-2, AUX 1-2	Select the controller input channel to be used as audio source of the selected program.
AMPLIFIER INPUT CHANNEL	Select the amplifier input channel to be used as local audio source.
CALL STATIONS	Select the call stations where the selected program will be listed in the menu.

UserMix Dialog
This dialog allows configuring audio routings (e.g. background music) in the system.



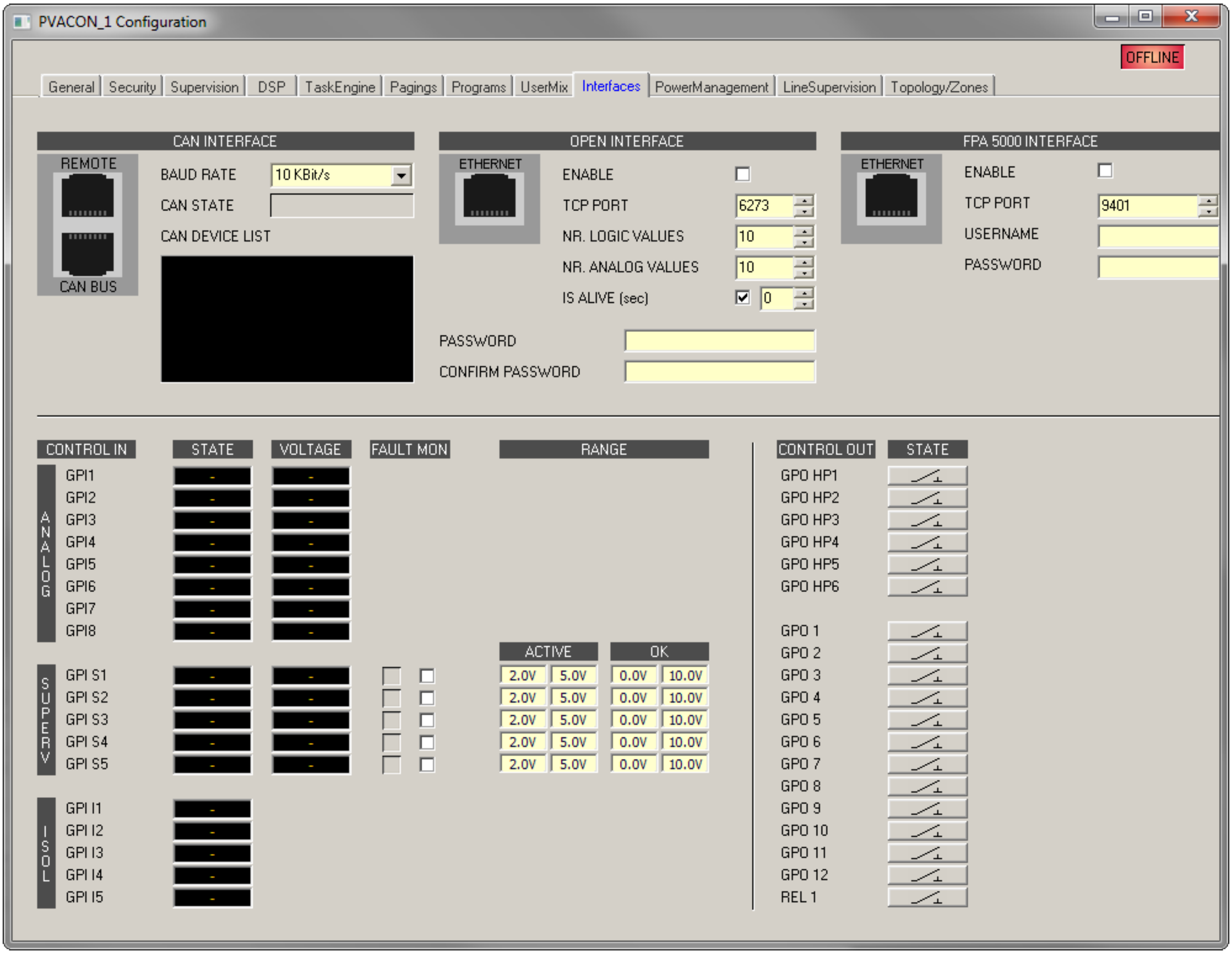
Element	Description
CONDITION	The condition that starts the background music, e.g. a switch connected to a GPI of the device.
INVERT	Set the checkbox to invert the condition that starts the back ground music.
SOURCE	The source of the background music.
DESTINATIONS	The destinations (zones or groups) of the background music.
REQUEST	Indicates the current status (active or inactive)

Element	Description
DELETE	Press the DELETE button to delete the entry selected in the list.
UPDATE	Press the UPDATE button to apply the settings in the lower section of the dialog to entry selected in the list.
NEW	Press the NEW button to create a new background music using the settings in the lower section of the dialog and adds it to the list.

Element	Description
CONDITION	
FILTER and condition list	Select the condition to start background music from the list. Enter a string (e.g. GPI) in the text field FILTER to list only the conditions containing this string.
SOURCE	
EXISTING	Select EXISTING to select an existing source for background music from the dropdown menu.
NEW/EDIT	Select NEW/EDIT to edit the settings of the source.
LABEL	Enter the name of the background music.
INPUT CHANNEL	Select the audio input channel for the background music.
DESTINATIONS	Select the zones or groups of the background music.

Interfaces Dialog

The Interface window allows configuring the different interfaces located on the rear panel of the device. All REMOTE CAN BUS and CONTROL PORT settings can be made in here. Configuring the Ethernet interface is done under Network Settings in the General window.

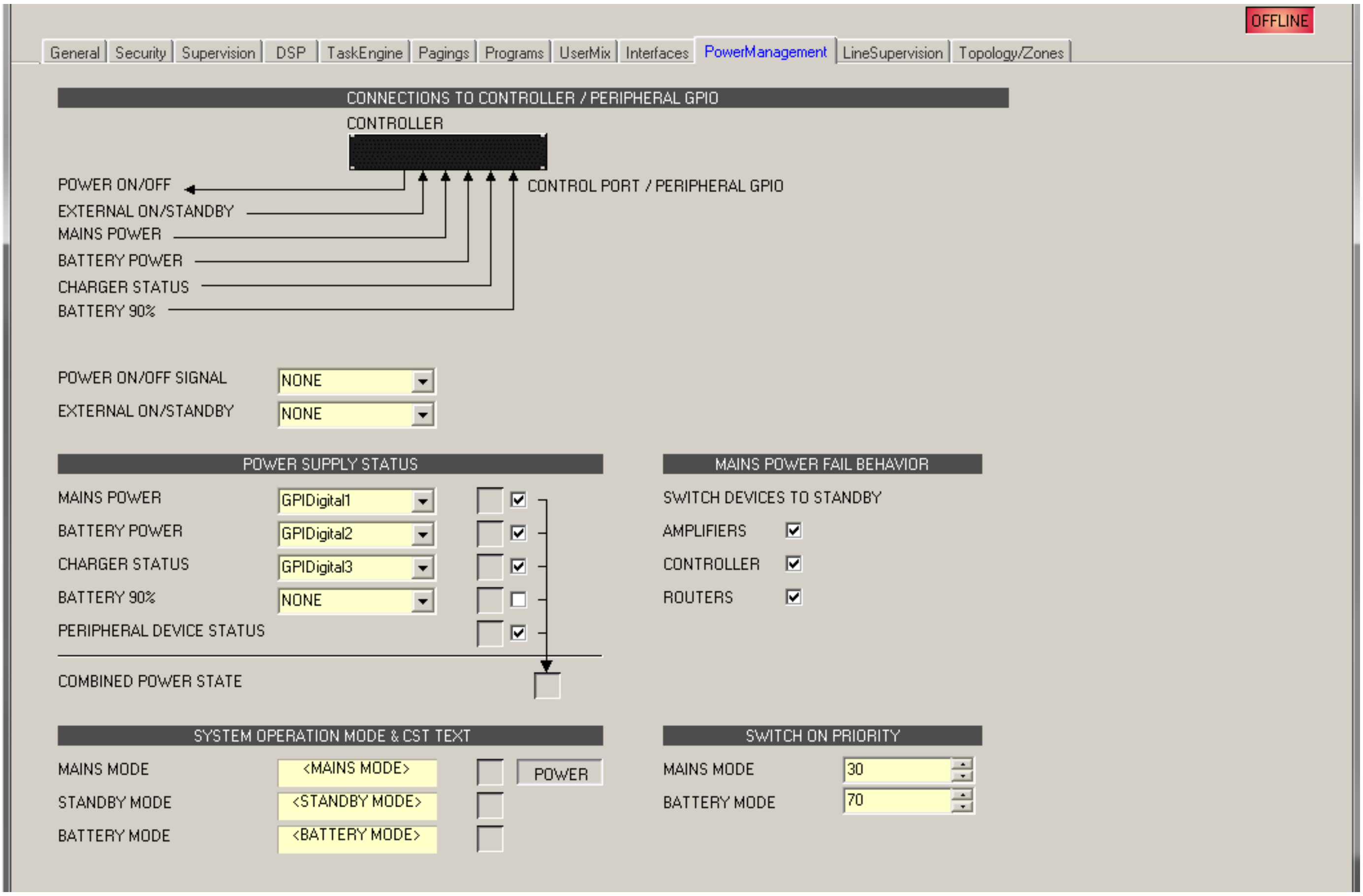


Element	**Description**
CAN INTERFACE	
BAUD RATE	Transmission rate of the CAN-Bus. All devices on the CAN-Bus must be set to one common transmission rate. **HINT: Editing the CAN BAUD RATE setting is possible in offline mode only.**
CAN STATE	Displays the current CAN-Bus status. Possible indications are: BUS OK, Bus Heavy, Bus Off.
CAN DEVICE LIST	Lists the connected devices.
OPEN INTERFACE	
ENABLE	Set the checkbox to activate the ASCII control protocol of the device.
TCP Port	TCP port of the ASCII control protocol. The default port is 6273.
NUMBER OF LOGIC VALUES	Enter the number of logic values of the task engine to be available via the ASCII control protocol.

NUMBER OF ANALOG VALUES	Enter the number of analog values of the task engine to be available via the ASCII control protocol.
IS ALIVE PERIOD (s)	Enter the is alive period of the ASCII control protocol in seconds.
PASSWORD	If password protection of the ASCII control protocol is required, enter the password here. Repeat the password in the CONFIRM PASSWORD field. Go online (write) to set the password in the device. **HINT: Editing the password setting is possible in offline mode only.**
FPA 5000 INTERFACE	
ENABLE	Set the checkbox to activate the connection between a FPA 5000 4000 and the PMX-4CR12 via Ethernet.
TCP Port	TCP port of the FPA 5000 interface. The default port is 9401.
USERNAME	Enter the username defined in the FPA.
PASSWORD	Enter the password defined in the FPA.
CONTROL IN	
STATE	Displays the control inputs' current state.
VOLTAGE	Displays the control inputs' current voltage.
FAULT MON	Set the checkbox of supervised control inputs to activate the supervision.
ACTIVE	Set the upper and lower bound (voltage) of the ACTIVE state of the supervised control inputs.
OK	Set the upper and lower bound (voltage) of the OK state of the supervised control inputs.
CONTROL OUT	
STATE	It is possible to manually change the condition of the control outputs (normally open contact / normally closed contact).

PowerManagement Dialog

The Power Management dialog allows configuring the standby mode of the device in detail.



Element	Description
POWER ON/OFF SIGNAL	Select the GPO contact or the virtual TE value for signaling the controllers' operating mode. In standby mode the GPO is open.
EXTERNAL ON/STANDBY	Select the digital GPI or virtual TE value to be used for switching to standby mode.
POWER SUPPLY STATUS	
MAINS POWER	Select the digital GPI or virtual TE value that is used for signaling „mains power OK". Set the checkbox to monitor this status.
BATTERY POWER	Select the digital GPI or virtual TE value that is used for signaling „battery power OK".Set the checkbox to monitor this status.
CHARGER STATUS	Select the digital GPI or virtual TE value that is used for signaling „charger status OK".Set the checkbox to monitor this status.
BATTERY 90%	Select the digital GPI or virtual TE value that is used for signaling „battery status at least 90%". Set the checkbox to monitor this status.
PERIPHERAL DEVICE STATUS	Set the checkbox to monitor this status.
COMBINED POWER STATE	This LED is green, if all selected power supply status are OK.
SYSTEM OPERATION MODE & CST TEXT	

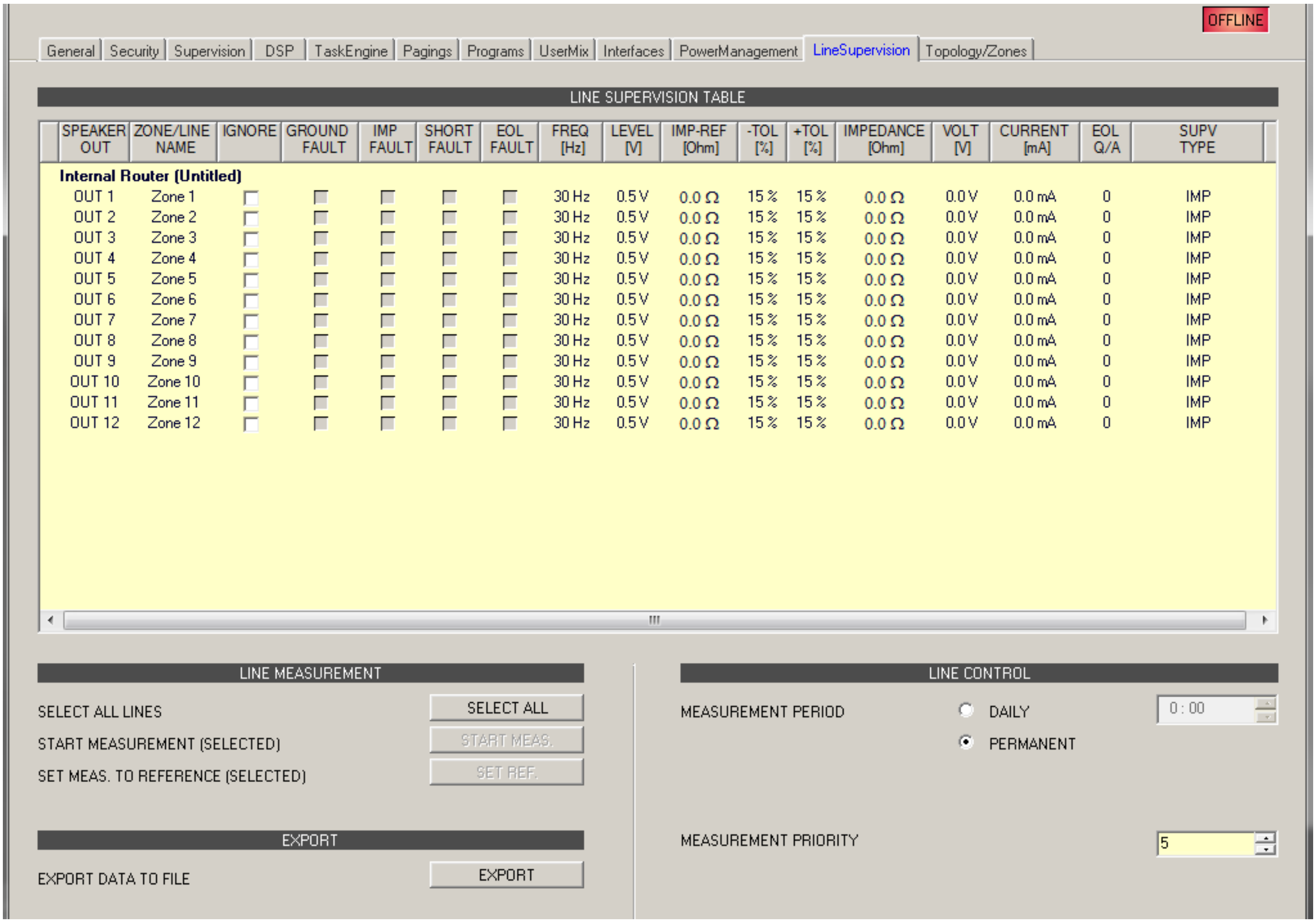
MAINS MODE	If mains power is used for running the system, the Controller is in MAINS MODE and the LED lights green. You can edit the name of this mode in the text field. Press the POWER button to power on/off the device.
STANDBY MODE	If the system is in STANDBY MODE, this LED lights green. You can edit the name of this mode in the text field.
BATTERY MODE	If battery power is used for running the system, the Controller is in BATTERY MODE and the LED lights green. You can edit the name of this mode in the text field.
MAINS POWER FAIL BEHAVIOR	
AMPLIFIERS	Select this option if Amplifiers should switch to standby mode if mains power fails.
CONTROLLER	Select this option if the Controller should switch to standby mode if mains power fails.
ROUTERS	Select this option if Routers should switch to standby mode if mains power fails.
SWITCH ON PRIORITY	
MAINS MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is available.
BATTERY MODE	Enter the minimum priority a signal (e.g. chime) must have to switch the system on, if the system is in standby mode and mains power is not available (battery mode).

Notice!
The “Power Calculator” tool can be used to calculate the power consumption of the system. The tool can be found in directory “/Tools” or can be requested from the IRIS-Net support team.

Notice!
The properties “Operating Mode” and “Standby LED” can be used for advanced power management configuration via the Task Engine, please refer to section Properties.

LineSupervision Dialog

The Line Supervision dialog allows configuring and control of the Controller line supervision. Line supervision can be done via line impedance measurement method or the End of Line method using PVA-1WEOL or Plena EOL modules. Please refer to the Controller manual for details about the different measurement methods.



Element	Description
SPEAKER OUT	System internal description of the zone or line.
ZONE/LINE NAME	Description of the zone or line.
IGNORE	Check this checkbox, if the result of the line measurement should be ignored. A error of this zone or line will not be indicated in the system. Regular measurements are carried out anyway. **HINT: If the checkbox is checked a short cut will not be indicated. If the zone is connected via a line relay, the relay will be deactivated.**
GROUND FAULT	This LED lights red, if a ground fault error has occurred.
IMP FAULT	This LED lights red, if the measured impedance is out of the tolerance range.
SHORT FAULT	This LED lights red, if there is a short cut at the zone or line (measured impedance value below 25% of reference value). In this case the system will not start calls or alarms in this zone or line. **HINT: If the zone is connected via a line relay, the relay will be deactivated when there is a short cut (short cut protection for other lines on the same amplifier).**
EOL FAULT	This LED lights red, if a EOL error has occurred.

FREQ [Hz]	Enter the frequency of the measurement signal.
LEVEL [V]	Enter the level of the measurement signal.
IMP-REF [Ohm]	Indicates the impedance reference value of the zone or line.
-TOL [%]	Maximum negative deviation of the impedance value of the zone or line from the reference value, given in percent.
+TOL [%]	Maximum positive deviation of the impedance value of the zone or line from the reference value, given in percent.
IMPEDANCE [Ohm]	Indicates the impedance value of the zone or line of the last successful measurement.
VOLT [V]	Indicates the voltage of the measurement signal of the last successful measurement.
CURRENT [mA]	Indicates the current of the measurement signal of the last successful measurement.
EOL Q/A	Indicates the quantity and addresses of the EOL modules in the zone or line.
SUPV TYPE	Select the supervision method used for the zone. Possible methods are: – IMP = impedance method – EOL = EOL method using EOL 8001 modules – PEOL = EOL method using Plena EOL modules
SELECT ALL	All zone or lines are selected.
START MEASUREMENT (SELECTED).	Starts the line measurement in all selected zones or line.
SET MEAS. TO REFERENCE (SELECTED).	Press this button to store the values of the last measurement as new reference values for the selected zones or lines.
EXPORT DATA TO FILE	All measurement data of the LINE SUPERVISION TABLE are exported to a csv file. Open the file in a spreadsheet for further processing.
DAILY	Set this checkbox, if a daily measurement should be done automatically. Enter the time the measurement should start.
PERMANENT	Set this checkbox, if line measurement should be done permanently.
PRIORITY	Priority of the line measurement signal.

The Line Supervision table is automatically generated from the available zones filled with default values.

Notice!
Use copy & paste to copy configurations from one element to another element in the line supervision table.

IMPEDANCE METHOD

The values of frequency, level and tolerance can be edited and adapted to the real conditions. To generate the reference values a first line measurement must be performed, the resulting measurement values are stored as reference values. The measurement of the lines and the comparison with the reference values is done automatically either permanently or every day at the scheduled time if the line is not busy. Each audio signal on the line interrupts the line measurement. The measurements will be continued automatically if the line is free again.

Notice!

Impedance reference values (IMP-REF) shall be measured and set for all used loudspeaker lines. The reference values are necessary not only if supervision type IMP is selected, but also for short circuit detection if supervision type EOL or PEOL is selected. The reference values are further needed for an impedance measurements triggered by an amplifier overload detection.

EOL METHOD

To enable the EOL supervision for a zone or line in the column EOL Q / A in the first line the number of EOL modules connected to the line must be entered, in the following lines the addresses of the modules must be entered. Enter 0 to disables the EOL method for the corresponding line.

Notice!

For power supply of the EOL modules a pilot tone is required, so the pilot tone generator of the power amplifier shall be activated.

Topology/Zones Dialog

The Topology/Zones dialog window allows configuration of Topologies and Zones. Zones are configured in a topology, each zone can be selected to be member of a Group.

IRIS-Net offers basic validation checks if the connections are valid. If a rule is not followed correctly, the connection line turns red.

The following rules apply:

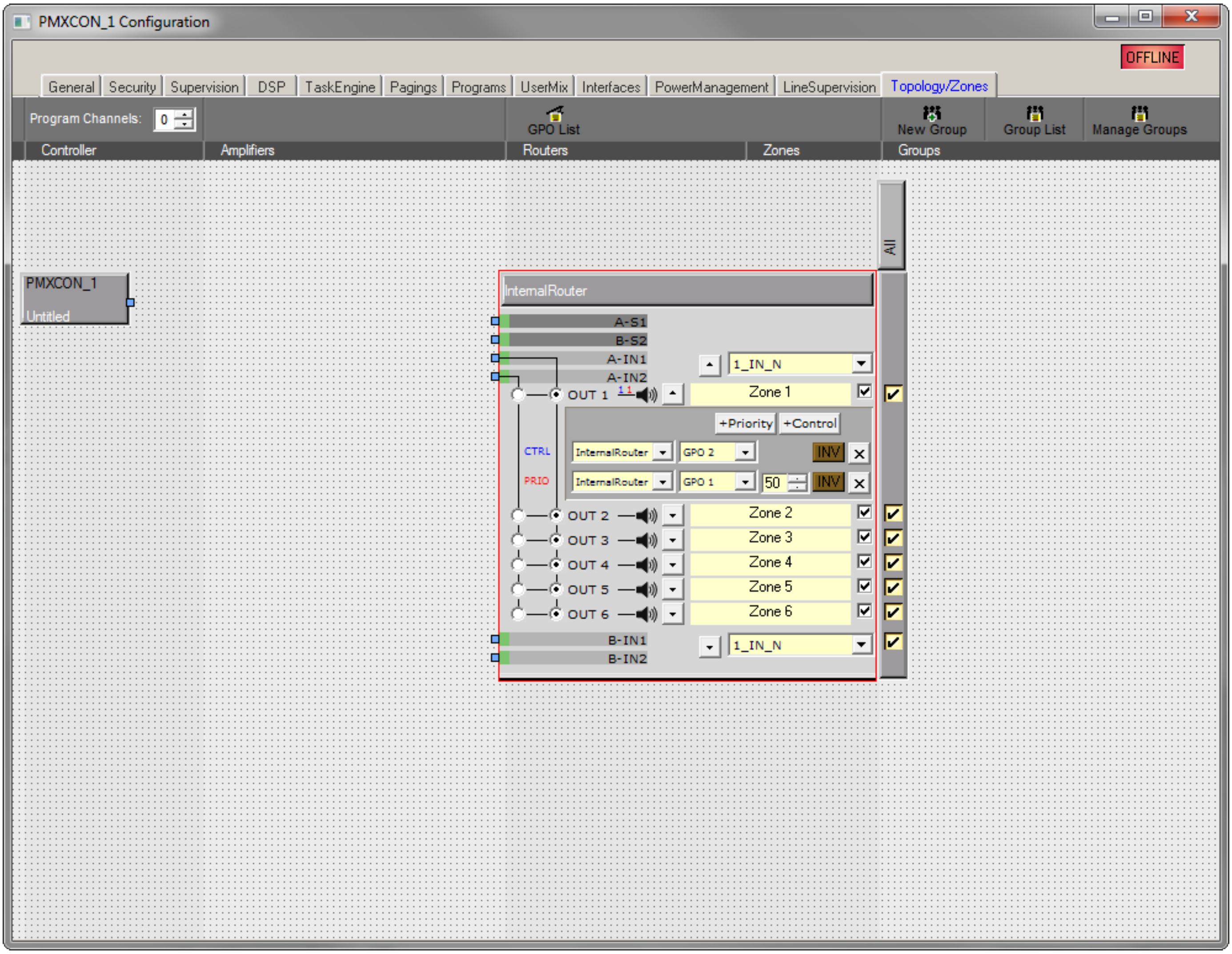
– One amplifier output can be connected to one or more router cluster inputs in parallel please refer to Line topology. Only identical inputs of router clusters can be connected in parallel, e.g. if a amplifier output is connected with the AMP IN1 input of one router cluster, the amplifier output can only be connected to AMP IN1 inputs of other router clusters. The same is valid for the router cluster input AMP IN 2. So it is not allowable to connect different types of router cluster inputs to one amplifier output.
– One amplifier output can always be used for one topology only. E.g. if a amplifier output is connected to a router cluster set to 1-in-N topology, the amplifier output can not be connected to a router cluster using another type of topology (e.g. 2-in-N).
– If the two output channels of an amplifier are used for a 2-in-N or Program/Call topology and the outputs are connected to more than one router cluster, the outputs have to be connected to the same input (e.g. always AMP IN 1) of the router clusters.
– If a 2-in-N or Program/Call topology is selected, going online is not possible if the identical amplifier output is connected to both inputs.
– Do not connect an amplifier output to a “regular” input of a router cluster (AMP IN 1, AMP IN 2) and a spare amplifier input (S1, S2,) at the same time.
– If router clusters with EOL8001 supervision are used, always connect one amplifier output to router clusters of one device only.

Notice!
If router clusters with EOL8001 supervision are used, switching router clusters within one device is possible. This allows creating 1-in-24 or 2-in-24 topologies with EOL 8001 supervision.

- If a 1-in-N topology is selected, the connections of the router cluster inputs (IN1 or IN2) have to be set correctly. E.g. every loudspeaker output has to be connected to a router cluster input, and the router cluster input has to be connected to a amplifier output.

For spare amplifier switching, the following rules apply:
- Automatic spare amplifier switching can be activated for every amplifier channel in the system. One spare amplifier channel is possible for an amplifier channel connected to the AMP IN 1 or AMP IN 2 input of a router cluster. The spare amplifier channel as to be connected to the S1 or S2 input of the same router cluster.
- If an amplifier output channel is connected to the inputs of more than one router clusters in parallel and should be backed up by a spare amplifier, the channel of the spare amplifier has to be connected to the same router clusters in parallel.

If a 2-in-N or Program/Call topology is selected and the amplifier should be backed up by a spare amplifier, two spare amplifier channels are required. It is not allowed to backup both channels of the topology with one spare amplifier channel only.



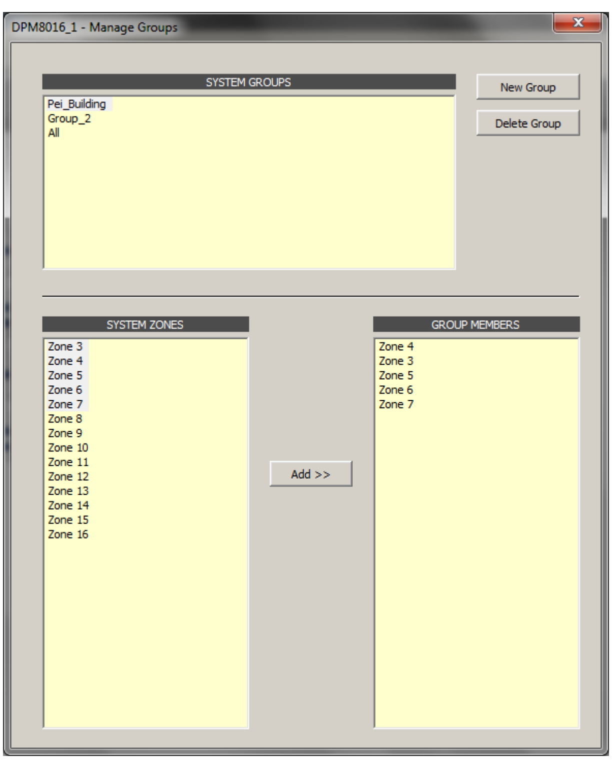
Icon bar

Element	Description
Program Channels	Select the number of program channels in the Controller.
GPO List	Click on this button to generate a CSV report of all GPOs configured in the system.
New Group	Click on this button to create a new Group. The “All” group, including all zones, is created automatically. For every new group the zones can be selected via the checkboxes in the group column.
Group List	Click on this button to generate a CSV report of all Groups configured in the system. The report includes the caption and object id of systems zones and the assignment of zones to system groups.
Manage Groups	Click on this button to open the Manage Group Dialog. This dialog allows to add or delete Groups and to add or remove Zones from a selected Group.

Element	Description
1_IN_N, 2_IN_N, PROG_CALL	Select the topology for the 2-in-6 cluster.
	Press this button to minimize or maximize the zones or relays dialog.
Zone 1	Enter a name for the zone.
+Priority	Press this button to add a priority relays to the zone. Note: Up to 2 priority relays can be configured in a zone.
+Control	Press this button to add a control relays to the zone. Note: Up to 2 control relays can be configured in a zone.
Device dropdown	Select the device to be used for controlling the control or priority relay.
GPO dropdown	Select the GPO (of the selected device) for controlling the control or priority relay.
50	This control allows setting the priority value of a priority relay.
INV	Press the INV button to invert the status of the control or priority relay.
X	Press this button to delete the corresponding priority or control relay.

Manage group dialog

This dialog allows to create, edit or delete groups. It is also possible to add or remove zones from a selected Group. To remove a zone from a group, select the zone in the GROUP MEMBERS section and press the delete button.



Element	Description
SYSTEM GROUPS	Lists all groups of the system.
New Group	Press this button to create a new group.

Element	Description
Delete Group	Press this button to delete the group selected in the SYSTEM GROUPS list.
SYSTEM ZONES	Lists all zones of the system.
Add >>	Adds the zones selected in the SYSTEM ZONES list to the group selected in the SYSTEM GROUPS list.
GROUP MEMBERS	Lists the zones currently included in the group selected in the SYSTEM GROUPS list.8

Properties

BUZZER CONTROL

The “PVACON_1.BuzzerControl” property of the PVA-4CR12 allows configuring the integrated buzzer. Following settings are available:

Value	Description
on	Buzzer is activated is a new error appears.
off	Buzzer is deactivated.
CST_1	Buzzer is activated if the call station (CST_1, CST_2,...) is not connected.

OPERATION MODE

The “PVACON_1.System.PowerManagement.OperatingMode” property allows setting the current operation mode of the PVA-4CR12 and connected devices. High priority signals prevent changing into standby mode. Following settings are available:

Value	Description
0	Switch PVA-4CR12 in standby mode
1	Switch PVA-4CR12 in operating mode

HINT: The mode of peripheral devices connected to the PVA-4CR12 is set automatically.

STANDBYLED

The Standby LED of the PVA-4CR12 lights, when the device is in standby mode. The corresponding property “PVACON_1.System.Info.StandbyLED” can be used to query the current mode.

Value	Description
0	PVA-4CR12 is in operating mode
1	PVA-4CR12 is in standby mode

PVA-4R24

The PVA-4R24 24 Zone Router is a zone extension for the PAVIRO system. The PVA-4R24 adds 24 zones, 20 GPIs, 24 GPOs and 2 control relays to the system and is controlled and supervised via the CAN bus by the PVA-4CR12 (Controller). Up to 20 external routers can be connected to one controller. One router can handle up to 4000 W speaker load. The maximum load of one zone is 500 W.

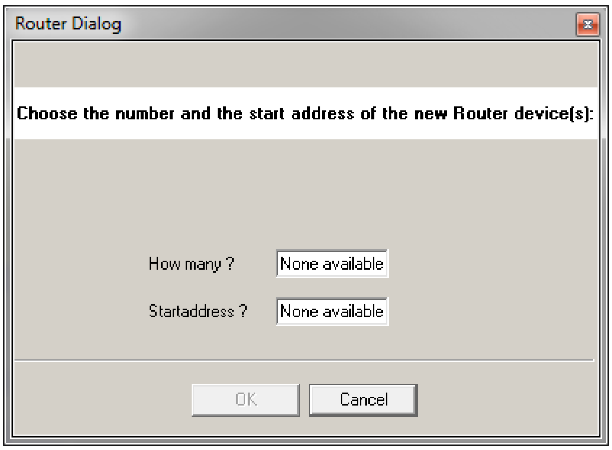
The zone indicator lights on the front indicate the current status of every zone:

– Green: Zone in use for non emergency purpose
– Red: Zone in use for emergency purpose

– Yellow: Zone fault detected
– Off: Zone in idle condition

PVA-4R24 Device

Start by creating a PVA-4R24 Device in your IRIS-Net project. Drag a PVA-4R24 from the Object Bar’s Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.

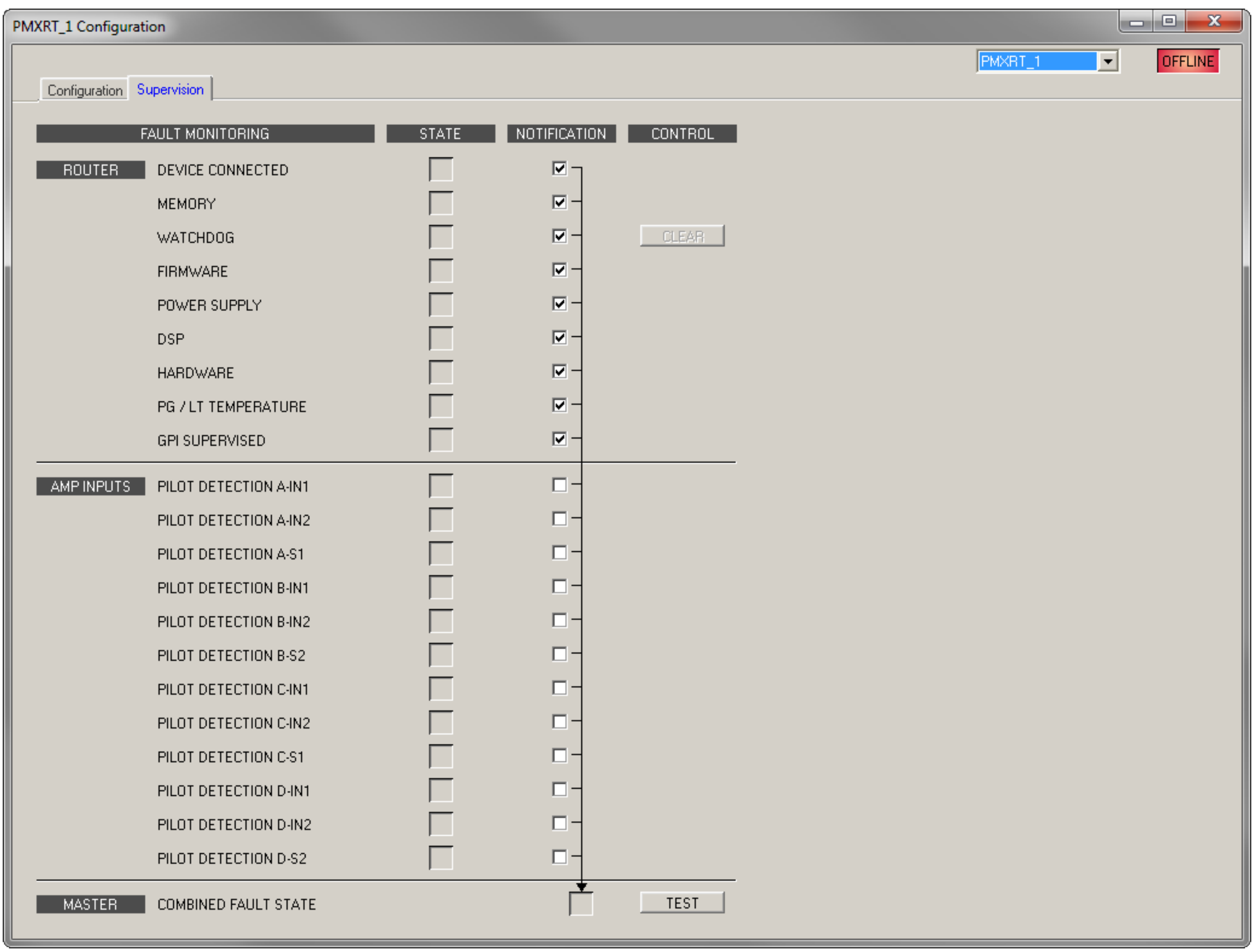
Configuration Dialog

Element	**Description**
MODEL	Indicates PVA-4R24, so you know the model of the device.
NAME	IRIS-Net internal device name of the Router.
FIRMWARE	Indicates the firmware version of the Router when on-line.
UPDATE	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
ADDRESS	Indicates the CAN address of the device.
FIND	Press the find button to activate the find function of the device.
OPERATING STATUS	Indicates the operating status of the Router.
CONTROL IN	
STATE	Displays the control inputs’ current state.

VOLTAGE	Displays the control inputs’ current voltage.
FAULT MON	Set the checkbox of supervised control inputs to activate the supervision.
ACTIVE	Set the upper and lower bound (voltage) of the ACTIVE state of the supervised control inputs.
OK	Set the upper and lower bound (voltage) of the OK state of the supervised control inputs.
CONTROL OUT	
STATE	It is possible to manually change the condition of the control outputs (normally open contact / normally closed contact).

Supervision Dialog

The Supervision window shows the condition of the PVA-4CR12. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a collected fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT indicator light on the front panel of the controller lights, the FAULT relay opens and a signal sound.
HOLD	Detected types of errors for which the checkbox HOLD is ticked are stored. Sporadic errors are indicated until the corresponding HOLD checkbox is unchecked.
LOG	
CONTROLS	

ROUTER

DEVICE CONNECTED	CAN connection between Controller and Router broken.
MEMORY	Memory error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
POWER SUPPLY	Error in the power supply of the device.
DSP	Error in the digital signal processing (DSP) of the device.
HARDWARE	Hardware error.
PG / LT TEMPERATURE	Temperature overload of the device.
GPI SUPERVISED	Voltage at supervised GPI out of range.

AMP INPUTS

PILOT DETECTION x-IN1	Missing pilot tone at input 1 of cluster A or B.
PILOT DETECTION x-IN2	Missing pilot tone at input 2 of cluster A or B.
PILOT DETECTION A-S1	Missing pilot tone at spare input 1 of cluster A.
PILOT DETECTION B-S2	Missing pilot tone at spare input 2 of cluster B.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

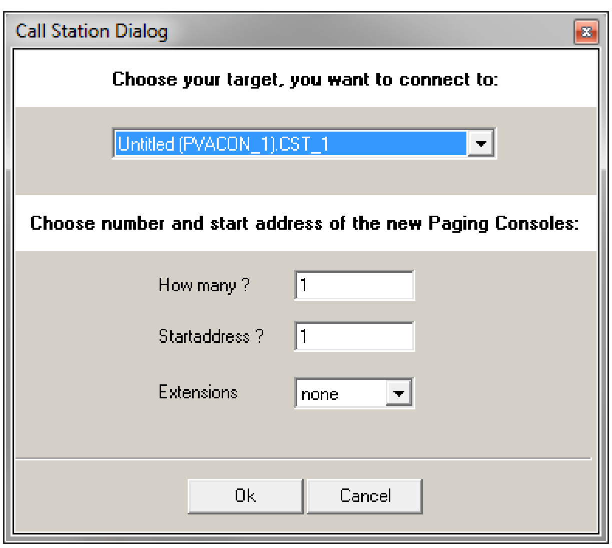
PVA-15CST

The PVA-15CST is a call station for the PAVIRO system. As standard, the call station has a gooseneck microphone with pop shield and permanent monitoring, a total of 20 buttons, an illuminated LC display, and an integrated loudspeaker. The call station can be modified to suit the user's requirements by connecting up to five PVA-20CSE call station extensions, each with 20 customizable selection buttons.
Other properties:

- Five menu/function keys (pre-programmed) – one green or one yellow indicator light per button
- 15 selection buttons (customizable) – two indicator lights (green/red) per button
- Label with transparent covering – the label can be changed at any time
- Can be used as a standing or desk/rack flush-mounted device
- Internal monitoring with error logging – complying with all relevant national and international standards
- Easy configuration – use of the Configuration Wizard or IRIS-Net software

PVA-15CST Device

Start by creating an PMX-15CST device in your IRIS-Net project. Drag an PMX-15CST from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

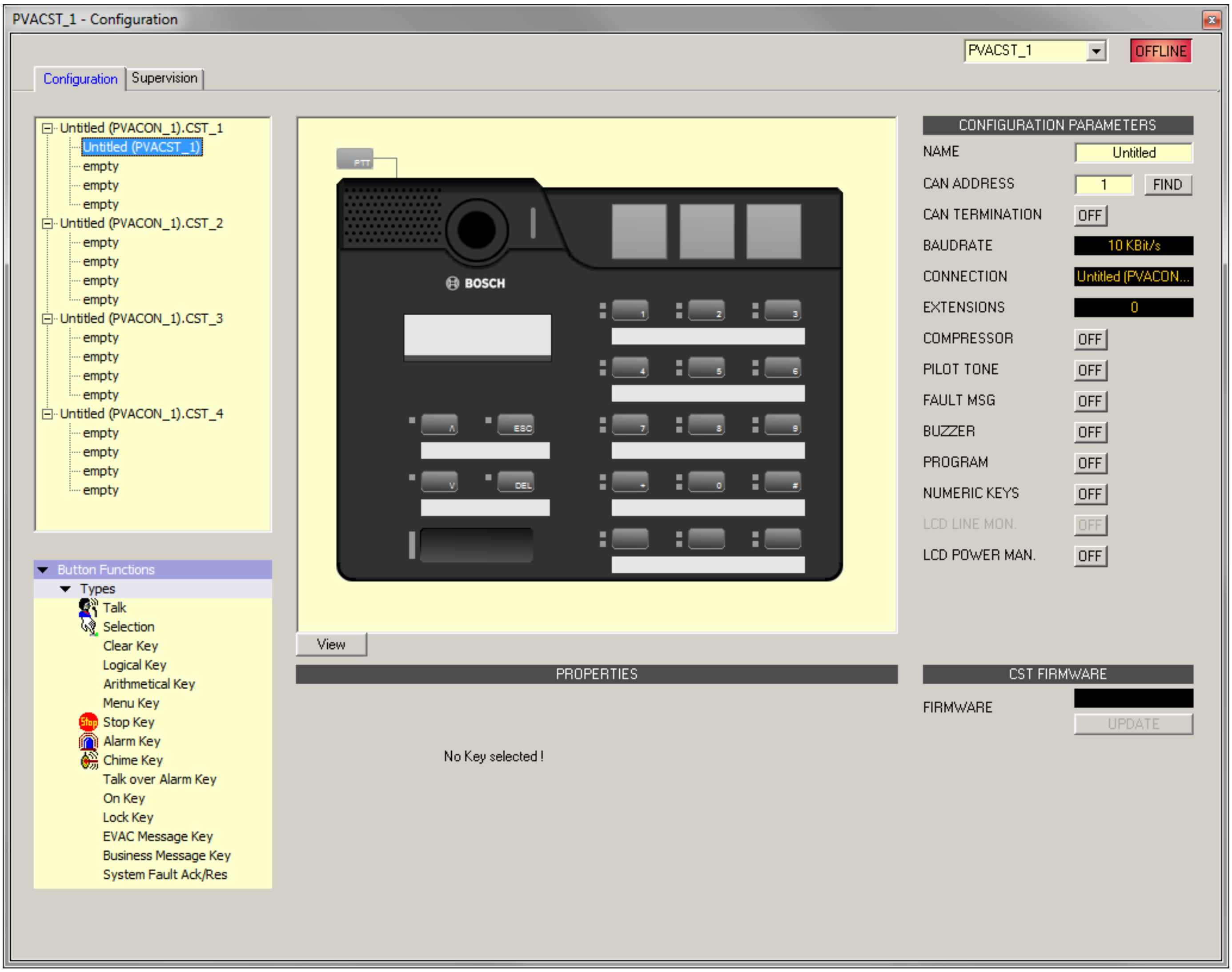


Select the call station bus the device is connected to.
Specify the desired number of devices, the address of the call station and number of call station extensions (it is not possible to add extensions to a call station kit). Click on the OK button to accept these settings.
The specified number of Call Stations will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

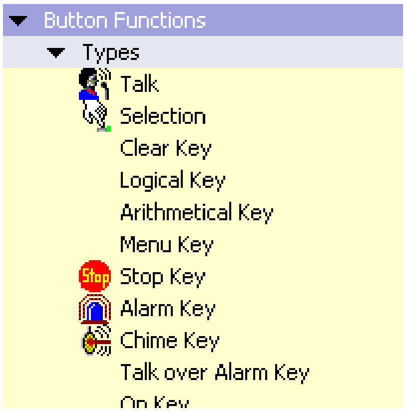
Double clicking on a Call Station device icon opens the configuration dialog window. Double clicking on a device for the first time will open the Configuration dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the Call Station device icon.

Configuration Dialog

This page allows making basic settings and retrieve information, for example of button functions, network settings, device name, firmware version, etc.

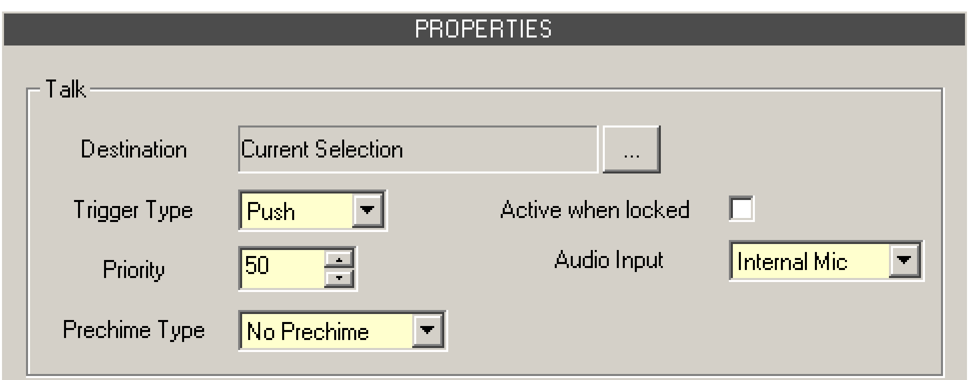


Element	Description
	When there are several call stations connected to the CST busses of the Controller you can select the call station (kit) to configure here.

	Select the desired button type and drag it from this dialog box onto the button of a call station or a call station extension. Detailed information about different types of buttons is provided on the following pages.
ONLINE OFFLINE	The Online / Offline indicator signals whether the call station is included in the network or off-line. The red OFFLINE indicator signals that the corresponding call station is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding call station is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
NAME	Name of the device.
CAN ADDRESS	Displays and lets the user enter the CAN address of the call station. Left-click in the field and enter the desired address in the range from 1 to 16. The entered value is adopted by pressing RETURN. The entered address has to match the setting in the call station's menu and may only exist once. When adding new call stations to an IRIS-Net project, CAN addresses are automatically assigned in ascending order.
FIND	When pressing this button, the backlight of the call station's LCD screen blinks regularly in quick succession. The status indicator of the call station Device in IRIS-Net blinks at the same time. This function serves for checking communication and for identification or search of a call station in a larger system.
CAN TERMINATION	Press this button (ON) to activate the internal termination resistor of the CAN bus in the call station.
BAUDRATE	The baud rate of the call station.
CONNECTION	Name of the Controller the call station is connected to.
EXTENSION	Number of Call Station extensions.
COMPRESSOR	Press this button (ON) to activate the internal compressor of the call station.
PILOT TONE	Press this button (ON) to activate the pilot tone supervision of the call station. **HINT: When using the pilot tone supervision only one call station can be connected to a CST bus.**
FAULT MSG	Press this button (ON) if error messages should be indicated in the LC-display of the call station.

BUZZER	Press this button (ON) if errors should be signaled via the integrated buzzer.
PROGRAM	Press this button (ON) if the Program Assignment menu should be accessible in the LC-display of the call station.
NUMERIC KEYS	Press this button (ON) to allow numeric entry of zone numbers.
LCD POWER MAN.	Press this button (ON) to indicate power management states in the display of the call station.
	Switching between the following views of a call station and (if existing) call station extensions: – Scroll View – Overall View – Selective View
FIRMWARE	Indicates the firmware version of the Call Station when on-line.
UPDATE	Press this button to update the firmware of the call station. **NOTE: The default password for the firmware update is “0000”.**

Talk
A switch of the type "Talk" allows configuring a TALK button. Specific Zones and/or Groups can be pre-selected for this key. Pressing the button on the call station automatically selects the Zones and/or Groups in which the spoken message is being heard.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Trigger (triggers a function)
Priority	Select the button's priority (0 to 9).
Audio Input	Select one of the following audio sources for the announcement: – Internal Mic – External Mic – External Line

Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Prechime Type	Select the desired type of pre-gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – No Prechime – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime

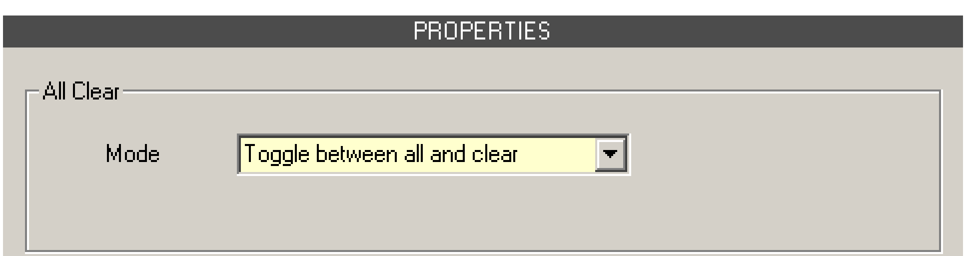
Selection

A switch of the type "Selection" allows configuring a SELECT button. Pressing the button on the call station selects the Zones and/or Groups that have been configured here.

Element	Description
Destinatio n	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.

Clear Key

A switch of the type "Clear Key" allows configuring an ALL/CLEAR button. Pressing the button on the call station selects or deselects all Zones and/or Groups.



Element	Description
Mode	Select the function that is to be executed when pressing the button on the call station: – Toggle between all and clear = Each press of the button alternately selects or deselects all Zones and/or Groups. – Select All = Pressing the button selects all Zones and/or Groups of the whole system. – Deselect All = Pressing the button deselects all Zones and/or Groups.

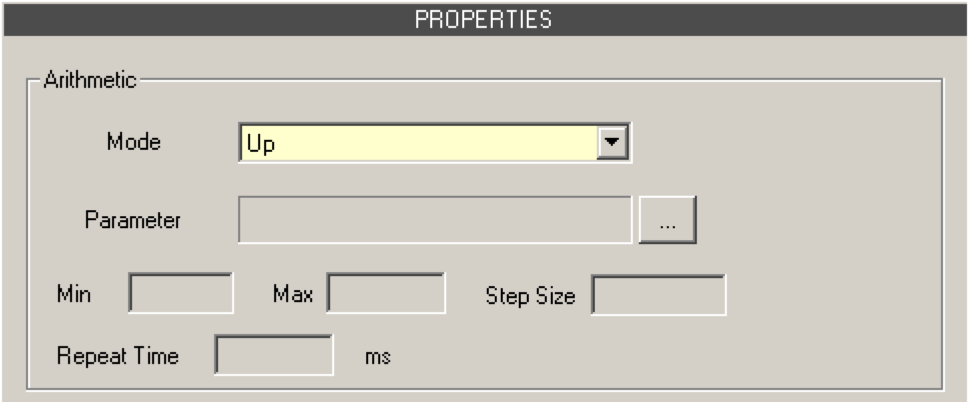
Logical Key

A switch of the type "Logical Key" allows setting the value of a logic variable (0 or 1). Pressing the button on the call station sets the logic variable to the desired value. The adjacent LED is operated according to the resulting parameter.

Element	Description
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Set Value = sets the value of the logic variable to "1". It remains "1", even after the button is being released. – Reset Value = sets the value of the logic variable to "0". It remains "0", even after the button is being released. – Push = sets the value of the logic variable to "1", but only as long as the button is being pressed. – Toggle = inverts the value of the logic variable each time the button is being pressed. – LED only = indicates the value of the logic variable, the value is not changed by the button
On	Select the LED of the button that should indicate the value „1“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Off	Select the LED of the button that should indicate the value „0“ of the logic variable: – Primary LED (green/red) – Secondary LED (yellow) – None
Parameter	The logic variable whose value is being changed.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Arithmetical Key

A switch of the type "Arithmetical Key" allows changing the value of a numerical variable. Pressing the button on the call station either increases or decreases the value of the numerical variable.



Element	Description
Mode	Select the desired parameter change that is to be executed when pressing the button on the call station: – Up = increases the value of the numerical variable – Down = decreases the value of the numerical variable
Parameter	The numerical variable whose value is being changed.
Min	The lower limit of the value range. Using the "Down" mode decreases the value of the numerical variable till down to this value.
Max	The upper limit of the value range. Using the "Up" mode increases the value of the numerical variable till up to this value.
Step Size	Lets the user enter the step width by which the value is to be changed when pressing the button on the call station.
Repeat Time	Lets the user enter a value for the time interval in milliseconds after which (when keeping the button pressed) the numerical value is being changed by the set step width art any one time. Entering "0" changes the value only once, even when keeping the button pressed over a longer period of time.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

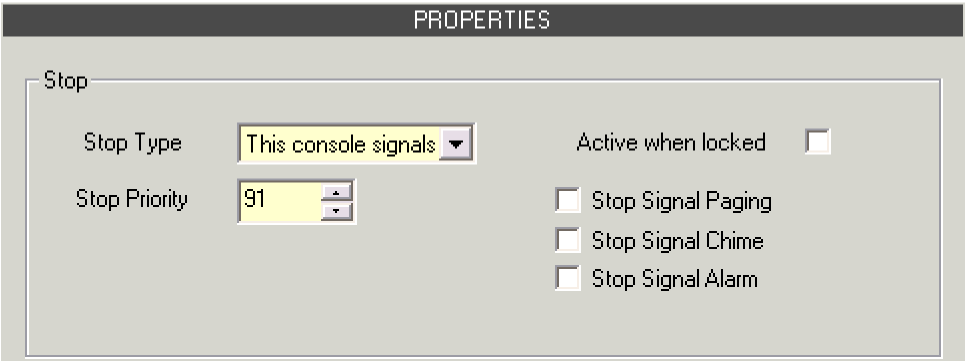
Menu Key

A switch of the type "Menu Key" displays the menu on the LCD screen of a call station.

Element	Description
Jump to	Select the position in the menu structure that is to be displayed

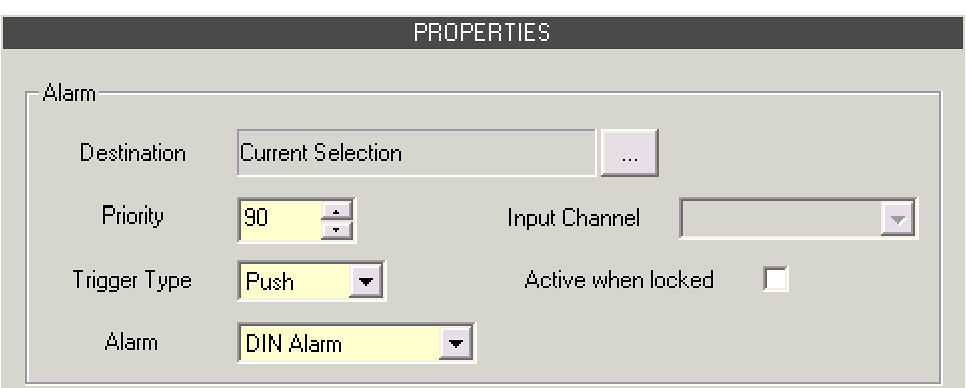
Stop Key

A switch of the type "Stop" allows canceling a process that is currently running on the system.



Element	Description
Stop Type	Select the function that is to be executed when pressing the button on the call station: – This concsole signals (local actions) = stops only the types of actions that have been launched from this specific call station – System signals = stops all selected types of actions system-wide, even if they have not been launched from this specific call station
Stop Priority	Select the maximum priority for the signals that will be stopped when pressing the button on the call station.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Stop Signal Paging	Pressing the button on the call station stops pagings.
Stop Signal Chime	Pressing the button on the call station stops chimes.
Stop Signal Alarm	Pressing the button on the call station stops alarms.
Stop Signal Text	Pressing the button on the call station stops signal texts.

Alarm key
A switch of the type "Alarm" allows starting an Alarm on the system.

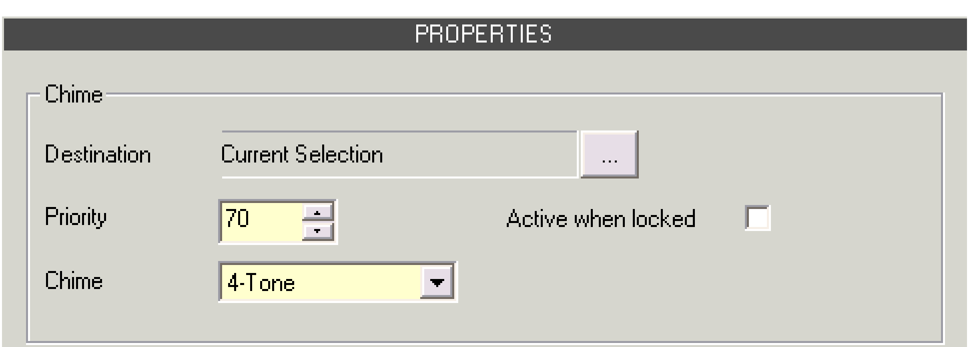


Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.

Priority	Select the alarm priority (0 to 100).
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger (triggers a function)
Alarm	Select the desired signal that is to be used for alarming: – Extern – DIN Alarm – Slow Whoop – Siren – Two-Tone Alarm – Telephone Alarm – Ship Alarm 1 – Ship Alarm 2 – Ship Alarm 3 – Ship Alarm 4 – Ship Alarm 5 – Ship Alarm 6 – Ship Alarm 7 – Ship Alarm 8 – Ship Alarm 9 – Ship Alarm 10 – Ship Alarm 11 – Ship Alarm 13 – Ship Alarm 14 – Ship Alarm 15 – Ship Alarm 16 – Ship Alarm 17
Input Channel	Enter the audio input at which the externally generated alarm signal is present.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Chime Key

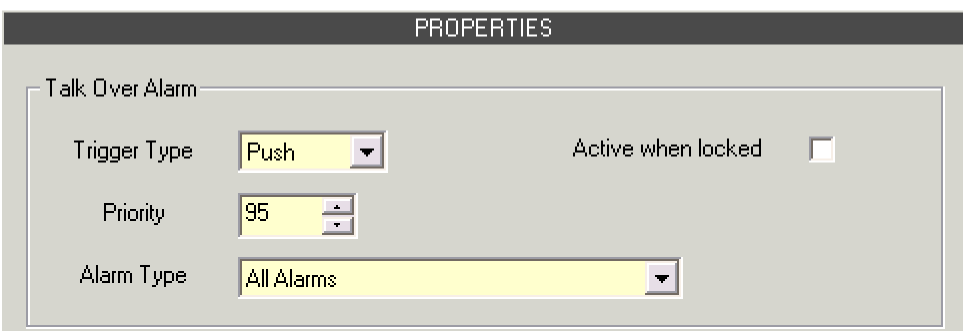
A switch of the type "Chime Key" allows the launch of a gong (chime) signal in the system.



Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the chime priority (0 to 100).
Chime Type	Select the desired type of gong (chime) signal. The list includes default signals and chime signals uploaded to the MM-2 module (if available). Following default signals are available: – 1-Tone – 2-Tone – 3-Tone – 4-Tone – 2x2-Tone – 2-Tone Pre-Chime
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Talk over Alarm Key

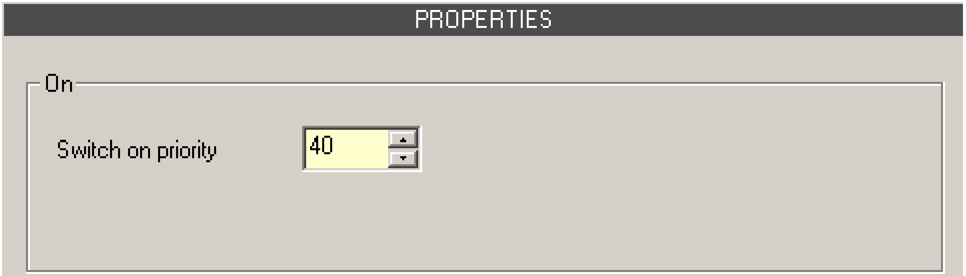
A switch of the type "Talk over Alarm Key" allows making an announcement during an alarm. During the announcement the alarm signal is off, and is started again after the announcement.



Element	Description
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states)
Priority	Select the priority (0 to 100) of the announcement. Must be higher than the alarm signal priority.
Alarm Type	Select the alarm type.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

On Key

A switch of the type “On” allows switching the PROMATRIX 8000 system on or off (standby) using a button of the call station.



Element	Description
Switch on priority	Select the priority (0 to 100) of the button.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.

Lock Key

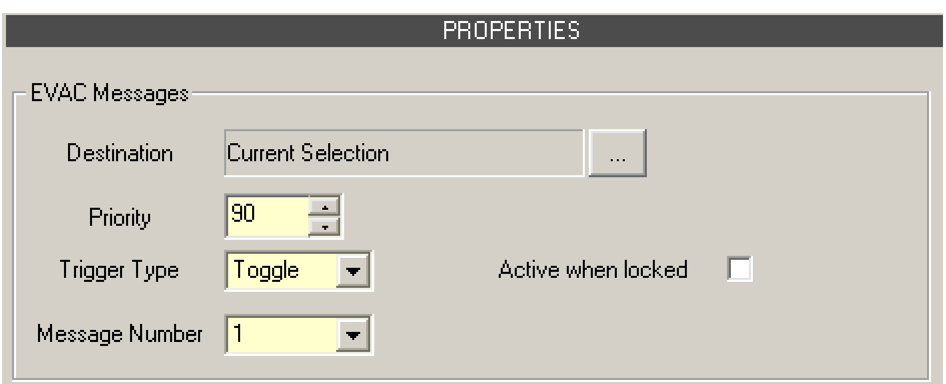
A switch of the type “Lock” allows locking the buttons of the call station. When assigned to a selection button the password set in the Security tab of the Controller has to be entered at the call station..

Notice!

If a button should stay active even if the call station is locked, the „Active when locked“ checkbox of this button has to be selected.

EVAC Message Key or Business Message Key

A switch of the type "EVAC Message Key" or “Business Message Key” allows starting a prerecorded message of type EVAC ore Business Message from the Message Manager.

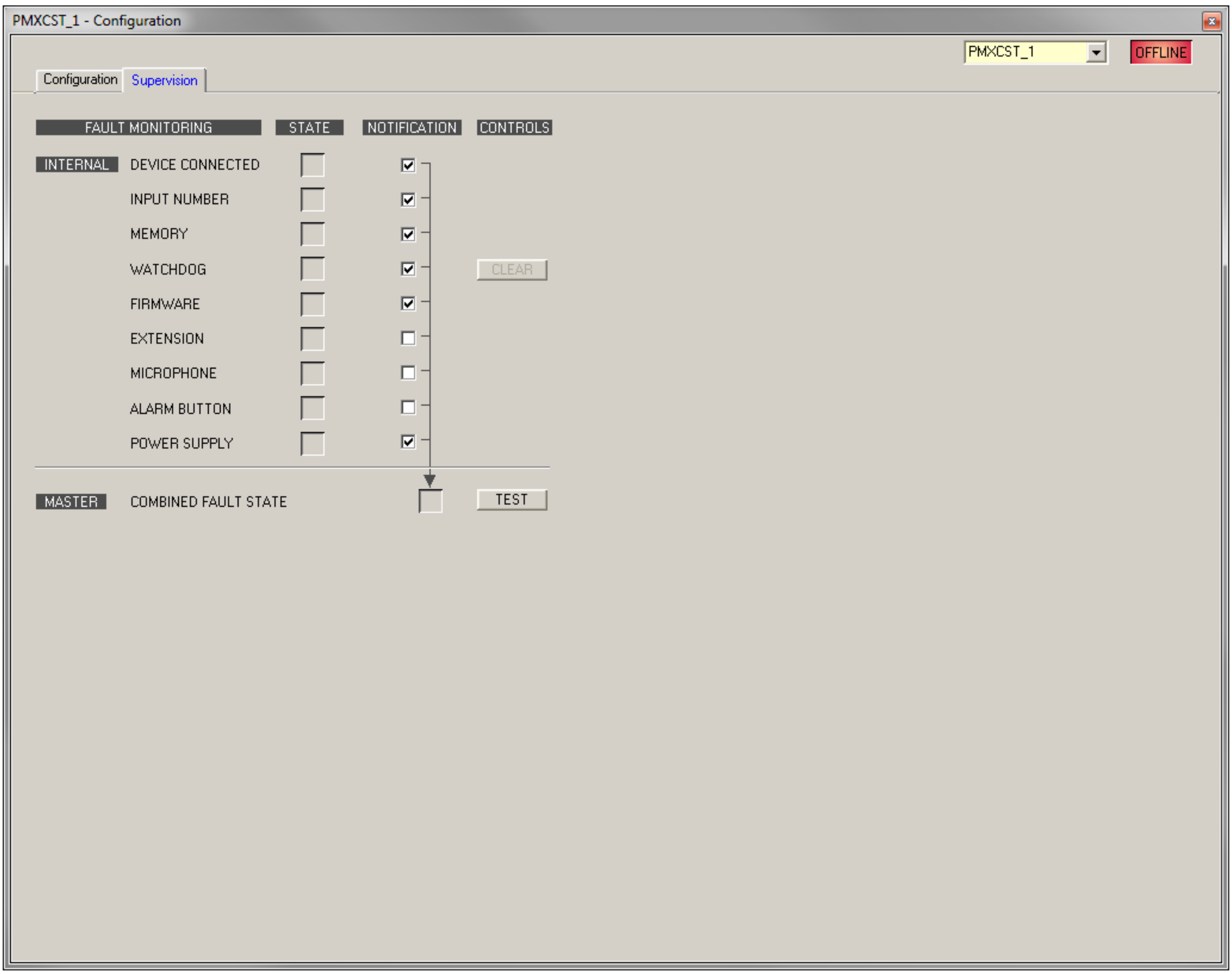


Element	Description
Destination	Clicking onto the button „..." opens the Destinations Dialog for selecting desired Zones and/or Groups.
Priority	Select the priority (0 to 100) of the message.
Trigger Type	Select the desired functionality for a button on a call station; available are: – Push (pushbutton) – Toggle (switches between two states) – Trigger

Message Name	Select the message by name.
Active when locked	Selecting this checkbox allows the user to press the button even though the call station has been locked.
Loop	Select this checkbox to automatically repeat the selected message.

System Fault Ack/Res
A switch of the type “System Fault Ack/Res” allows to acknowledge or reset a system fault that is indicated at the call station. This type can be assigned to the DEL button only.

Supervision Dialog



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox NOTIFICATON is ticked, the COMBINED FAULT STATE is set at the same time and the FAULT indicator light at the Call Station lights.
DEVICE CONNECTED	The CST bus connection between Controller and Call Station is broken.
INPUT NUMBER	The call station is not connected to the correct CST bus.
MEMORY	Memory error in the Call Station.
WATCHDOG + CLEAR	Watchdog error in the Call Station. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the Call Station is too old.
EXTENSION	The number of call station extensions is to high or the addresses of the extensions are not correct.
MICROPHONE	Microphone error in the Call Station.
ALARM BUTTON	Supervision fault of the alarm button or the key switch.
POWER SUPPLY	Power supply out of range.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

PVA-CSK

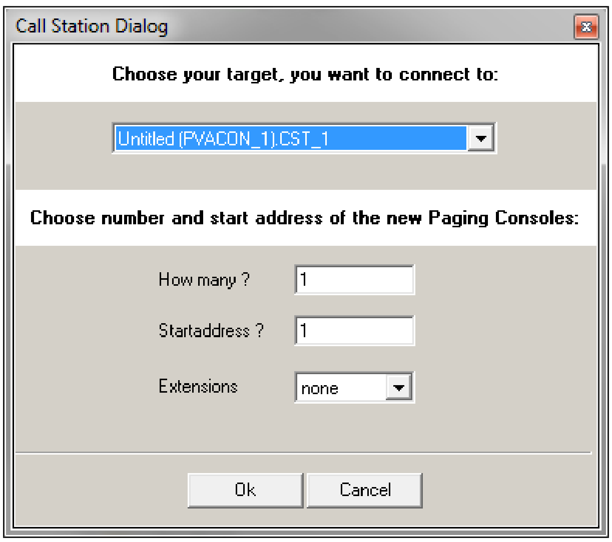
The PVA-CSK call station kit is a call station printed circuit board (PCB) for the PAVIRO system. The circuit board allows an application-specific call station to be installed, such as a fire department call station.
The call station kit is based on the call station, but has been optimized so that it is easy to adapt to different application areas. In addition to the stem microphone familiar from the PVA-15CST, a dynamic EMERGENCY microphone such as the DBB 9081 can also be connected. The call station kit equipped with an illuminated LC display (122 x 32 pixels). The call station has the following features:

– Possible to connect microphone with pre-amplifier and compressor/limiting switch
– Possible to connect five pre-programmed menu/function buttons
– Possible to connect up to 15 function and selection buttons, programmable button assignment
– Possible to connect up to three alarm buttons or key switches
– Possible to connect an external microphone or audio source
– Possible to connect a loudspeaker
– High-resolution LC display
– Comprehensive parameter settings menu on the actual call station
– Microphone and line monitoring
– Error message via LED and buzzer, and error text in the LC display
– Processor control of all functions
– Monitoring of the processor system via watchdog circuit
– Non-volatile FLASH memory for configuration data

The call station is processor-controlled, and equipped with extensive monitoring functions. Line monitoring for the CAN bus and for audio transmission allows line interruptions and short-circuits to be detected and indicated to the user. The microphone, PTT button, alarm button and key switch monitoring allows line interruptions and short-circuits to be detected and reported.
The call stations for the PAVIRO system can be configured quickly and easily using IRIS-Net. A graphical and dialog-based user interface allows the user to define all button functions, priorities, options, and other properties.

PVA-CSK Device

Start by creating a PVA-CSK device in your IRIS-Net project. Drag a PVA-CSK from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:

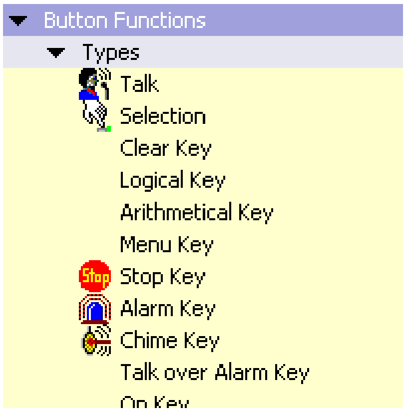


Select the call station bus the device is connected to.
Specify the desired number of devices, the address of the call station and number of call station extensions (it is not possible to add extensions to a call station kit). Click on the OK button to accept these settings.
The specified number of Call Stations will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.
Double clicking on a Call Station device icon opens the configuration dialog window. Double clicking on a device for the first time will open the Configuration dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the Call Station device icon.

Configuration

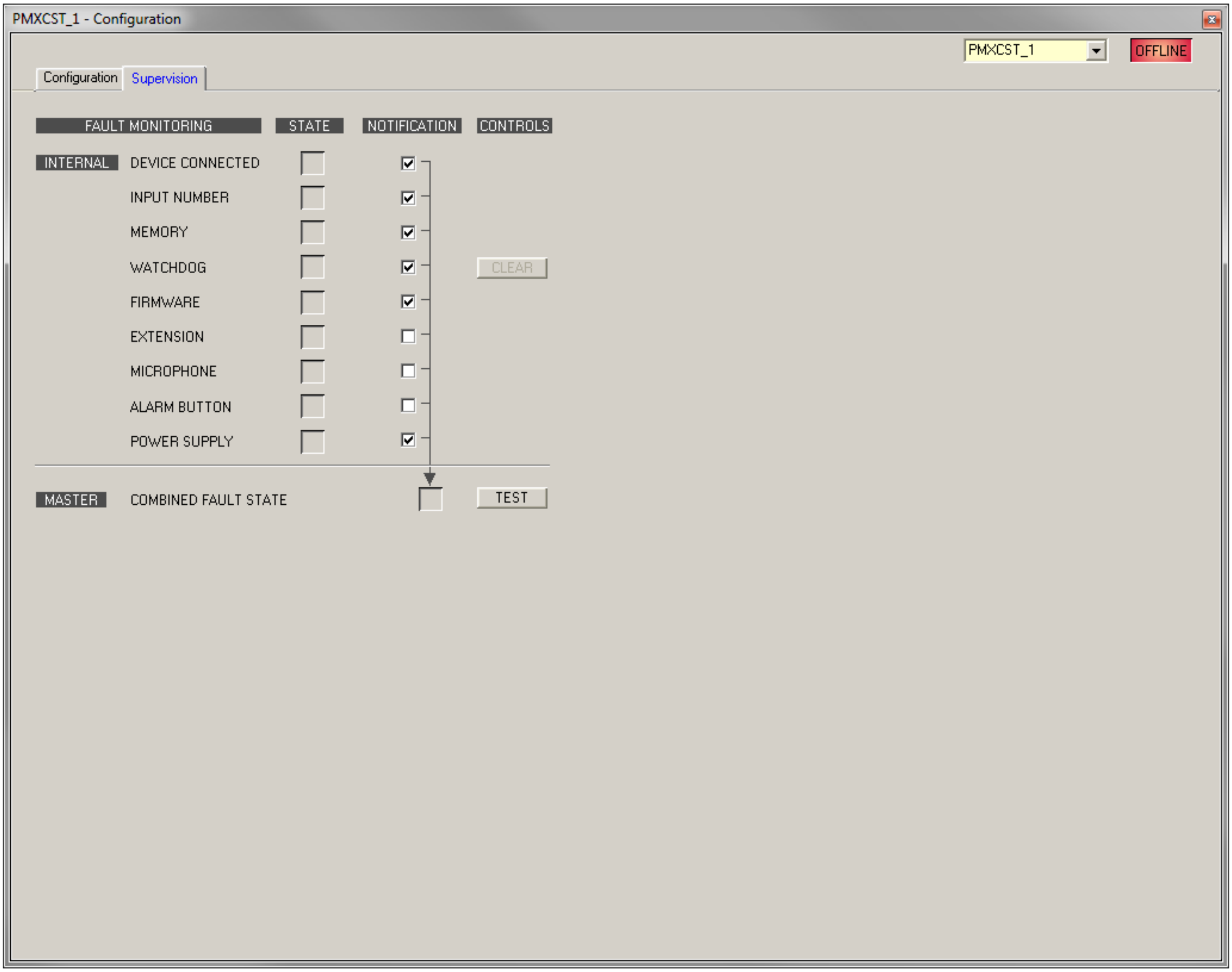
This page allows making basic settings and retrieve information, for example of button functions, network settings, device name, firmware version, etc.

Element	Description
	When there are several call stations connected to the CST busses of the Controller you can select the call station (kit) to configure here.

	Select the desired button type and drag it from this dialog box onto the button of a call station or a call station extension. Detailed information about different types of buttons is provided on the following pages.
ONLINE OFFLINE	The Online / Offline indicator signals whether the call station is included in the network or off-line. The red OFFLINE indicator signals that the corresponding call station is off-line and that therefore no communication is possible. The green ONLINE indicator shows that the corresponding call station is on-line and that sending and receiving data is possible. When on-line, any parameter changes are immediately transmitted and active.
NAME	Name of the device.
CAN ADDRESS	Displays and lets the user enter the CAN address of the call station. Left-click in the field and enter the desired address in the range from 1 to 16. The entered value is adopted by pressing RETURN. The entered address has to match the setting in the call station's menu and may only exist once. When adding new call stations to an IRIS-Net project, CAN addresses are automatically assigned in ascending order.
FIND	When pressing this button, the backlight of the call station's LCD screen blinks regularly in quick succession. The status indicator of the call station Device in IRIS-Net blinks at the same time. This function serves for checking communication and for identification or search of a call station in a larger system.
CAN TERMINATION	Press this button (ON) to activate the internal termination resistor of the CAN bus in the call station.
BAUDRATE	The baud rate of the call station.
CONNECTION	Name of the Controller the call station is connected to.
EXTENSION	Number of Call Station extensions.
COMPRESSOR	Press this button (ON) to activate the internal compressor of the call station.
PILOT TONE	Press this button (ON) to activate the pilot tone supervision of the call station. **HINT: When using the pilot tone supervision only one call station can be connected to a CST bus.**
FAULT MSG	Press this button (ON) if error messages should be indicated in the LC-display of the call station.

BUZZER	Press this button (ON) if errors should be signaled via the integrated buzzer.
PROGRAM	Press this button (ON) if the Program Assignment menu should be accessible in the LC-display of the call station.
NUMERIC KEYS	Press this button (ON) to allow numeric entry of zone numbers.
LCD POWER MAN.	Press this button (ON) to indicate power management states in the display of the call station.
	Switching between the following views of a call station and (if existing) call station extensions: – Scroll View – Overall View – Selective View
FIRMWARE	Indicates the firmware version of the Call Station when on-line.
UPDATE	Press this button to update the firmware of the call station. **NOTE: The default password for the firmware update is “0000”.**

Supervision



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.
NOTIFICATON	At the occurrence of a type of error for which the checkbox NOTIFICATON is ticked, the COMBINED FAULT STATE is set at the same time and the FAULT indicator light at the Call Station lights.
DEVICE CONNECTED	The CST bus connection between Controller and Call Station is broken.
INPUT NUMBER	The call station is not connected to the correct CST bus.
MEMORY	Memory error in the Call Station.
WATCHDOG + CLEAR	Watchdog error in the Call Station. This error type is logged according standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version of the Call Station is too old.

MICROPHONE	Microphone error in the Call Station.
PTT/ALARM BUTTON	Supervision fault of the PTT microphone or alarm button.
POWER SUPPLY	Power supply out of range.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

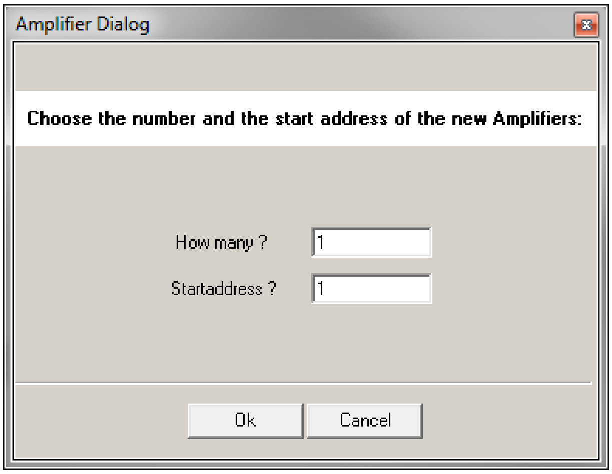
PVA-2P500

The PVA-2P500 class-D amplifier is a 2 × 500 W professional audio amplifier for evacuation purposes. It can be operated from both the mains and a DC supply. The output voltage is galvanically insulated and is constantly monitored for ground fault. An energy-saving mode and temperature-controlled fans reduce energy consumption and noise levels. The control and monitoring functions are performed via CAN bus. This amplifier is designed for operation in an emergency evacuation system. The amplifiers are usually controlled via a controller and configured using IRIS-Net.
The power amplifier has the following features:
- Floating 100 V or 70 V power outputs
- High efficient amplifier blocks in class-D technology
- Outputs idling and short circuit-protected
- Mains operation 120–240 V (50/60 Hz) and/or 24 V DC emergency backup
- Electronically balanced inputs
- Temperature monitoring function
- Pilot tone and ground fault monitoring function via PVA-4CR12 Controller or PVA-4R24 Router
- Processor control of all functions
- Monitoring of the processor system via watchdog circuit
- Non-volatile FLASH memory for configuration data
- Internal monitoring function
- Integrated audio relays
- Line monitoring function

The power amplifier is processor-controlled and equipped with extensive monitoring functions. Line monitoring for the CAN bus and for audio transmission allows line interruptions and short-circuits to be detected and indicated to the user.

PVA-2P500 Device

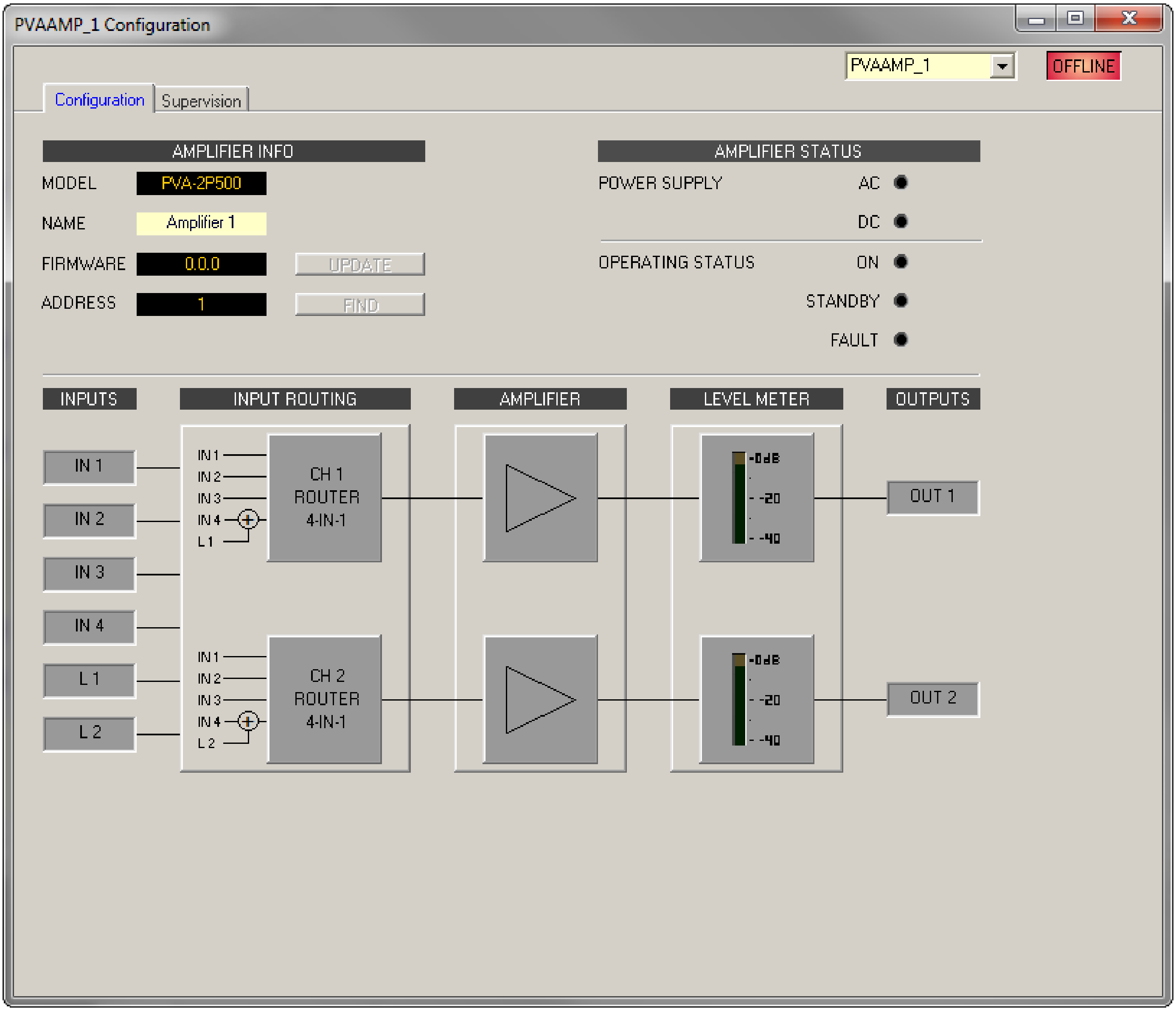
Start by creating a PVA-2P500 device in your IRIS-Net project. Drag a PVA-2P500 from the Object Bar's Devices category or from the Devices window into the worksheet (see also chapters: Devices and Configurations menu). The following dialog box appears:



Enter the required number of devices and select a communication interface. Click on the OK button to accept these settings. The specified number of devices will be created and displayed in the worksheet. Selected devices can be dragged around and repositioned at will. To select a device either click and drag the mouse to draw a rectangle around it or hold down the 'ctrl' key and click on the device. In either case a successfully selected device is shown with a red border around it.

Double clicking on an device icon opens the configuration dialog window. Double clicking on a device for the first time will open the General dialog box. Here, you can specify initial settings that are necessary for further configuration and communication. Additional configuration windows can be navigated to by clicking on the icons at the top of the window. However, as a basic rule, IRIS-Net will remember which window was used last and reopen to this window next time you double click on the device icon.

Configuration Dialog

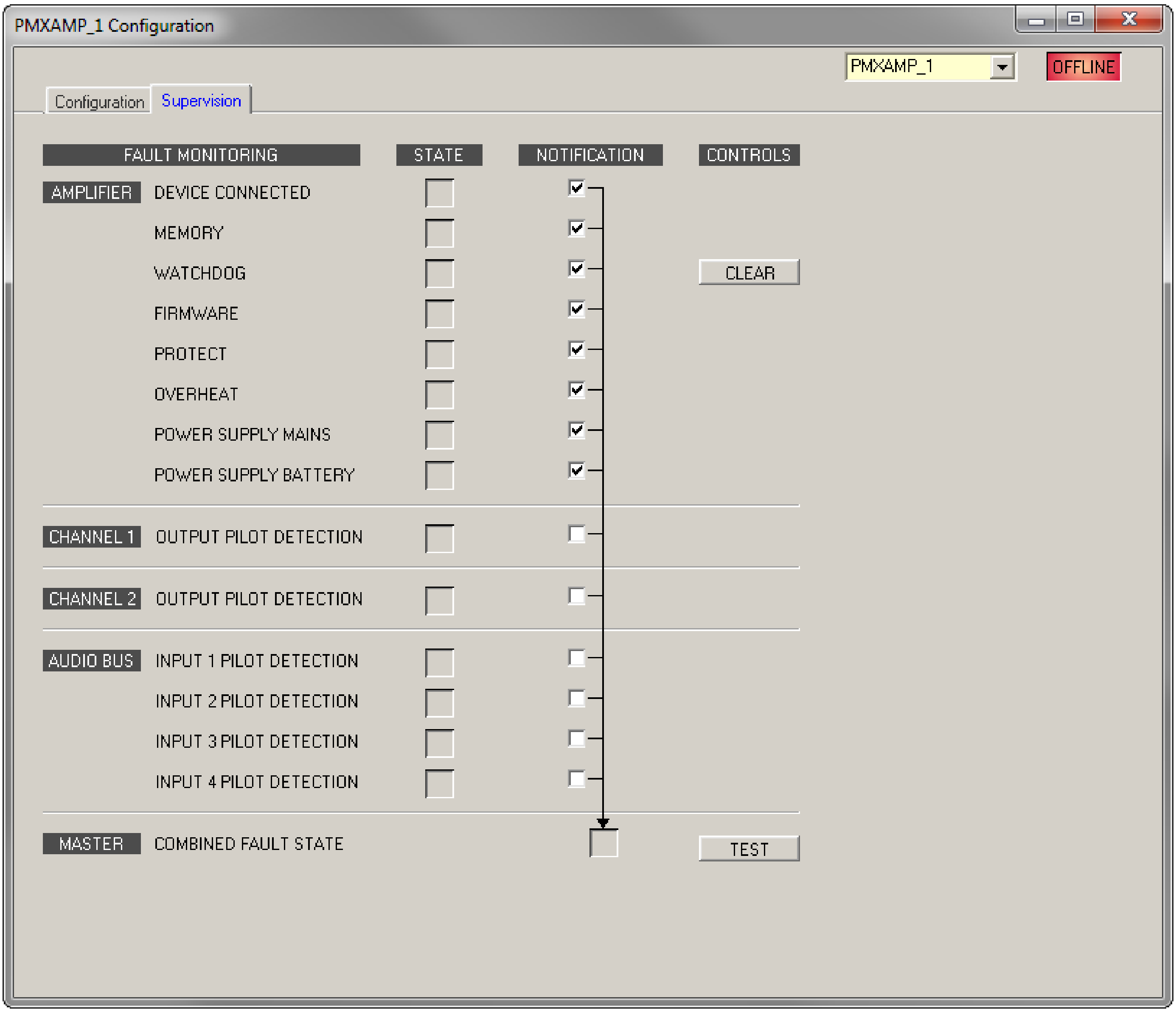


Element	Description
MODEL	Indicates PVA-2P500, so you know the model of the device.
NAME	IRIS-Net internal device name of the Amplifier.
FIRMWARE	Indicates the firmware version of the Amplifier when on-line.
	Opens the firmware update dialog. **NOTE: The default password for the firmware update is “0000”.**
ADDRESS	Indicates the CAN address of the device.
FIND	Press the find button to activate the find function of the device.
POWER SUPPLY	Indicates the status of the AC or DC supply voltage.

OPERATING STATUS	Indicates the operating status of the Amplifier.
LEVEL METER	The level meters of the two output channels indicate the signal level of the audio output signal.

Supervision Dialog

The Supervision tab shows the condition of the PVA-2P500. When on-line, all fault conditions are being indicated. It is possible to select for each type of error whether it is displayed in a combined fault message, buffered and/or indicated at the call station displays.



Element	Description
STATE	The current condition of each type of error gets indicated. Green means no error, red indicates that an error has been detected.

NOTIFICATON	At the occurrence of a type of error for which the checkbox DETECT is ticked, the COLLECTED ERROR STATE flag is set at the same time. Additionally the FAULT indicator light on the front panel of the controller lights, the FAULT relay opens and a signal sound.
CONTROLS	

Error types

DEVICE CONNECTED	CAN connection between Controller and Amplifier broken.
MEMORY	Memory error.
WATCHDOG	Watchdog error of the device. This error type is logged conforming to standards, press the CLEAR button to reset the error.
FIRMWARE	The firmware version is not compatible with the IRIS-Net version used. A firmware update is recommended.
PROTECT	Protect mode of amplifier activated.
OVERHEAT	Temperature overload of the device.
POWER SUPPLY MAINS	Error in the mains power supply of the device.
POWER SUPPLY BATTERY	Error in the battery power supply of the device.
OUTPUT PILOT DETECTION	Missing pilot tone at the amplifier output channel 1 or 2.
INPUT x PILOT DETECTION	Missing pilot tone at the amplifier input channels 1 to 4.

MASTER

COMBINED FAULT STATE	The FAULT indicator light on the front panel of the device lights at the occurrence of this type of error.
TEST	Manually setting or resetting an error.

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